Micronet
Faster and Easier Networks

User's Manual

IP Telephony Gateway

Model No.: SP5001A/S

http://www.micronet.info

Table of Contents

About this User's Manual

This user's guide gives hardware specifications and explains web configuration and command line configuration for the VoIP Telephony Gateway.

Specifications are subject to change without notice.

Online Upgrade

Please refer to http://www.micronet.info/ for additional support documentation.

General Syntax Conventions

Mouse action sequences are denoted using a comma. For example, click start, Settings, Control Panel, Network means first you click Start, Click or move the mouse pointer over Settings the click or move the mouse pointer over Control Panel and finally click (or double-click) Network.

"Enter" means to type one or more characters.

Predefined choices are in **Bold Arial** Font.

A single keystroke is in Arial font and enclosed in square brackets. **[Enter]** means the Enter.

For brevity's sake, we will use "e.g.," as shorthand for "for instance", and "i.e.," for "that is" or "in other words."

Safety Notes

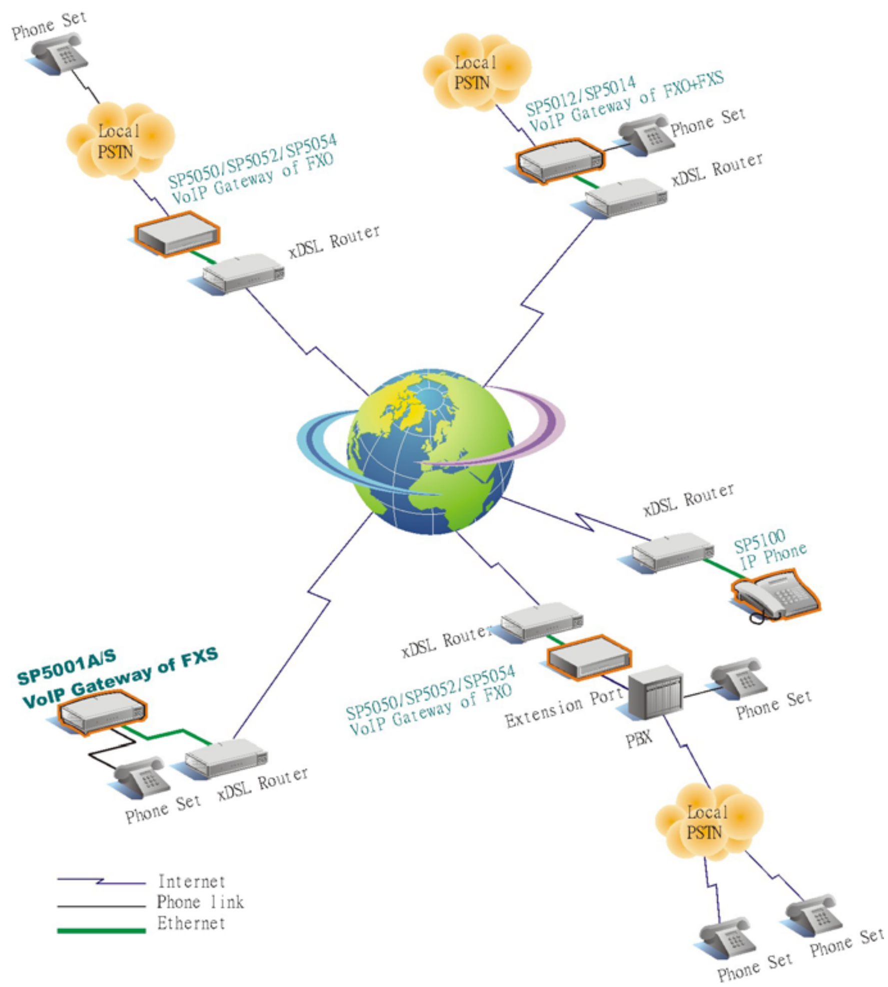
Use the external power supply that is included in the package. Other power supplies may cause damage to the phone, affect the behavior or induce noise.

1. Introduction

1.1. Overview

Micronet SP5001A/S FXS Gateway is designed to connect standard telephone devices to IP-based telephony networks, providing users with high-quality VoIP service. In addition, the 10/100M switch ports can offer network connection to co-located PC or other Ethernet-based devices. No need to prepare extra hubs or switches. SP5001A/S is an ideal solution for home users or small offices.

SP5001A/S is compliant with IETF RFC 3261 SIP standards, and has a built-in DHCP server to assign IP addresses automatically to your PCs, making configuration effortless. SP5001A/S can save the toll call expense and maximizes your broadband investment.



1.2. Features

- Compliant with IETF RFC 3261 SIP standards
- Provide 1 RJ-11 FXS port for phone set or fax machine
- Provide 3-port 10/100M Ethernet switch
- Provide advanced telephony features, such as call hold, call forward and call transfer.
- Support Proxy and Peer-to-Peer Mode
- Support FAX over IP (T.38)
- Support FSK and DTMF Caller ID
- Support Static IP, DHCP and PPPoE connection
- Built-in NAT for IP sharing
- Built-in DHCP Server
- TFTP/FTP firmware upgrade
- QoS : ToS (Type of Service)
- Support EMS (Element Management System)**

Audio feature

- Codec: G.711 a/μ-law, G.723.1 (6.3kbps), G.729A
- VAD (Voice Activity Detection)
- CNG (Comfort Noise Generate)
- G.168/165-compliant adaptive echo cancellation
- Dynamic Jitter Buffer
- Bad Frame Interpolation
- Voice/DTMF Gain Settings

Interface

- One 10/100 Base-T Ethernet RJ45 port for WAN
- Three 10/100 Base-T Ethernet RJ45 ports for LAN
- One RJ11 Telephone Port (FXS).
- DC 12V input.

System Management

WEB Interface, Telnet

Environment

Operating and storage Humidity: 10 to 90 % (Non-condensing)

Operational Temperature: 0 to 40 ℃

Storage Temperature: -10 to 50 ℃

Dimension & Weight : 190 x 124 x 37 mm, 320g

Certification
CE, FCC

1.3. Default Settings

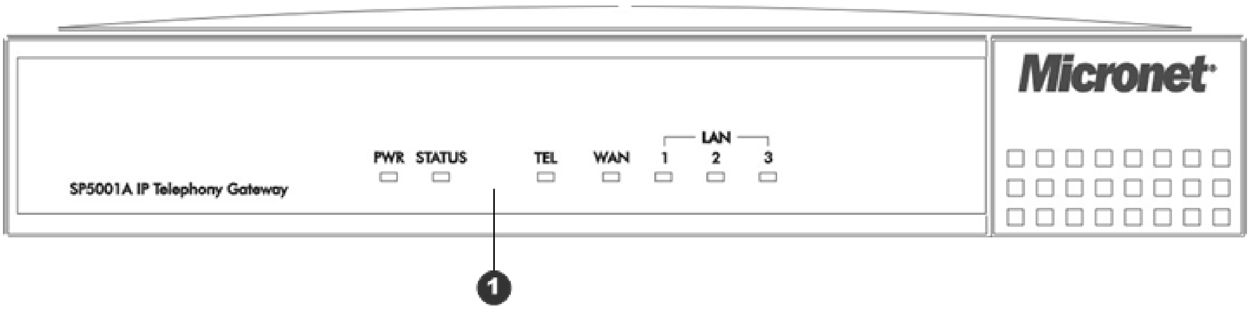
The following are the settings of the default profile

IP Parameters
WAN IP Address: 10.1.1.3   Subnet: 255.0.0.0   Default gateway: 10.1.1.254
LAN IP Address: 192.168.123.123

Telnet and Web Login Password
Login = root
Password = Null (default)

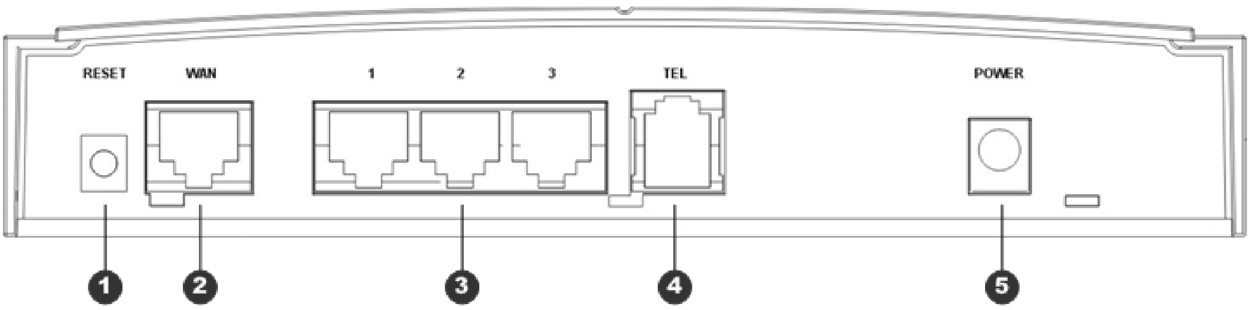
1.4. Appearance

Front Panel



1. LED Status Display

Rear Panel



1. Reset Button
2. RJ-45 WAN Port
3. RJ-45 LAN Ports
4. RJ-11 FXS Interface
5. Power Jack 12V DC

Note :
To restore the factory default configuration settings, press and hold the Reset button on the rear panel for more than 3 seconds. Release the Reset button and wait for the gateway to reboot.

LED Status Display:

LEDs	Functions	Status	Active	Description
PWR	Power	Green	On	The Power is on
			Off	The Power is off
STATUS	Status	Green	On	Gateway is under Proxy mode and registered to Proxy server successfully
			Off	Gateway is in Peer-to-Peer Mode
			Blinking	Gateway is in Proxy mode but no register, or Gateway is booting up
TEL	TEL	Green	On	The Telephone is Off-Hook
			Off	The Telephone is On-Hook
			Blinking	The gateway has Incoming Call
WAN	WAN	Green	On	WAN Port connected
			Off	WAN Port disconnected
			Blinking	WAN Port is transmitting or receiving data
LAN (1, 2, 3)	LAN Connection	Green	On	LAN Port connected
			Off	LAN Port disconnected
			Blinking	LAN Ports are transmitting or receiving data

Ethernet WAN Port:
Connect the Ethernet cable from gateway's WAN port to the ADSL or Cable modem Ethernet port.

Ethernet LAN ports:
Connect the Ethernet cable from gateway's LAN port to the Ethernet adapter in your computer.

TEL Port:
RJ-11 connector, FXS interface. To connect analog phone set or trunk line of PABX.

Power Jack:
12V DC Power supply.

2. Setting Up the Gateway

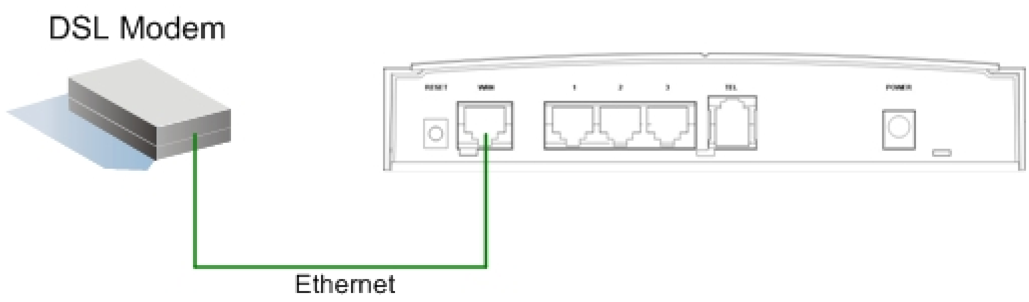
This chapter will describe the basic connection setup and configure the gateway via your web browser through a computer. It outlines how to connect your VoIP Gateway to the LAN and the WAN. In the case of connecting a Cable Modem you must connect the coaxial cable from your cable service to the threaded coaxial cable connect on the back of the cable modem.

2.1. Connecting the SP5001A/S

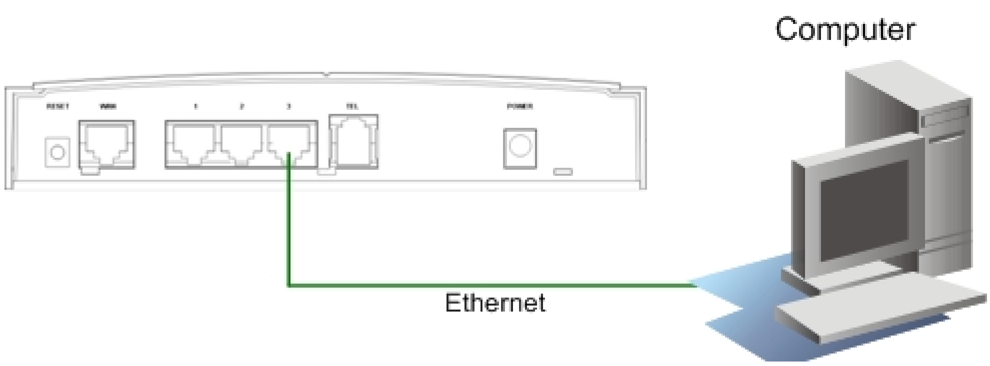
1. Turn off your computer

2. Turn off the DSL or cable broadband modem

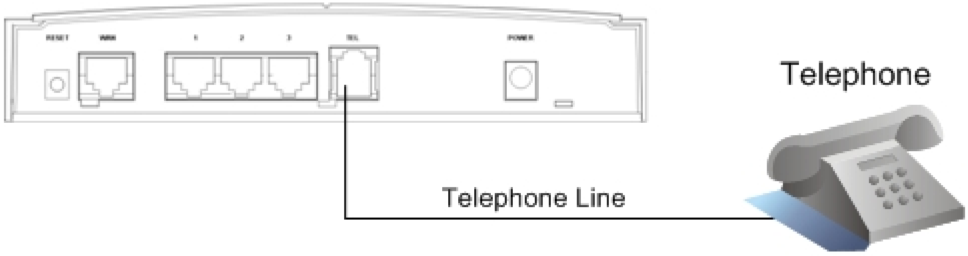
3. Connect the Ethernet cable from WAN port to the ADSL or Cable modem Ethernet port.



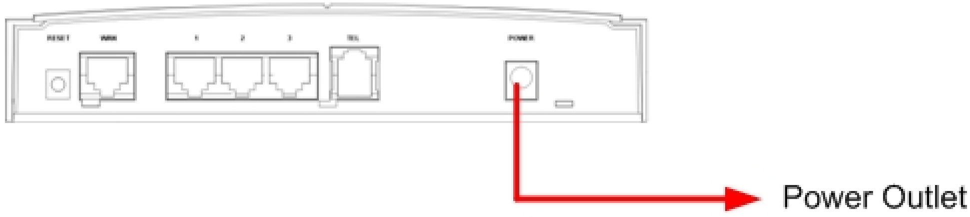
4. Connect the Ethernet cable from LAN port to the Ethernet adapter in your computer



5. Connect the telephone handset to the TEL port (FXS port)

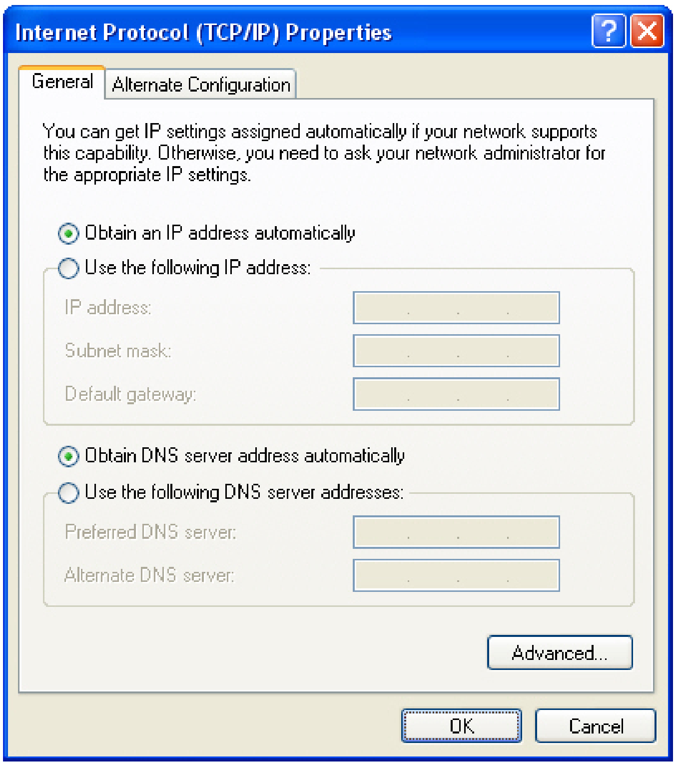


6. Connect the power adapter to the gateway and plug it in to a power outlet. It takes about 40 seconds to boot up completely



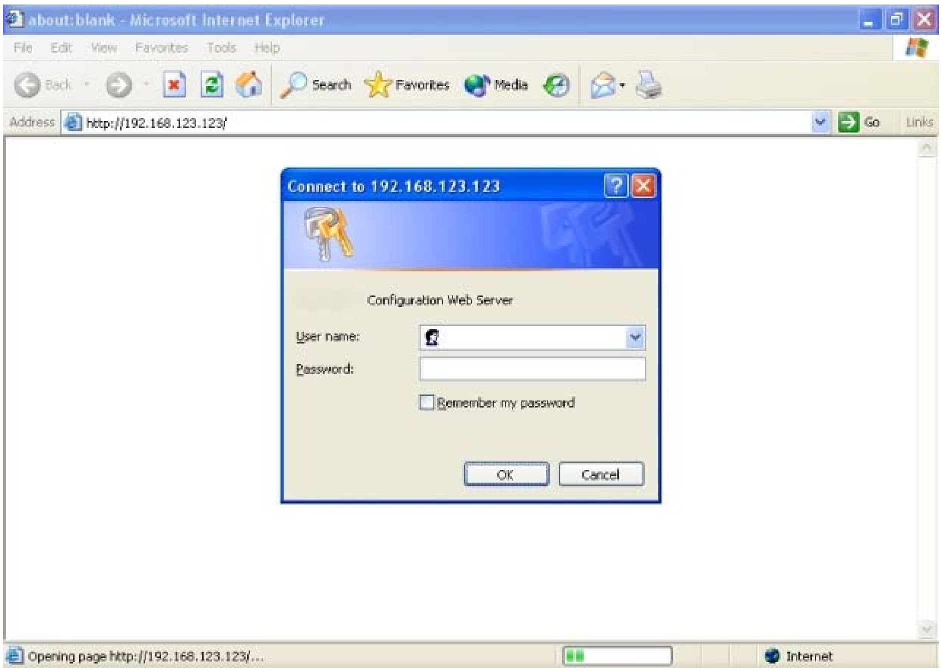
7. Power on your computer and DSL modem

8. Configure your PC network adapter to set to automatically get its TCP/IP configuration from the SP5001A/S via DHCP.

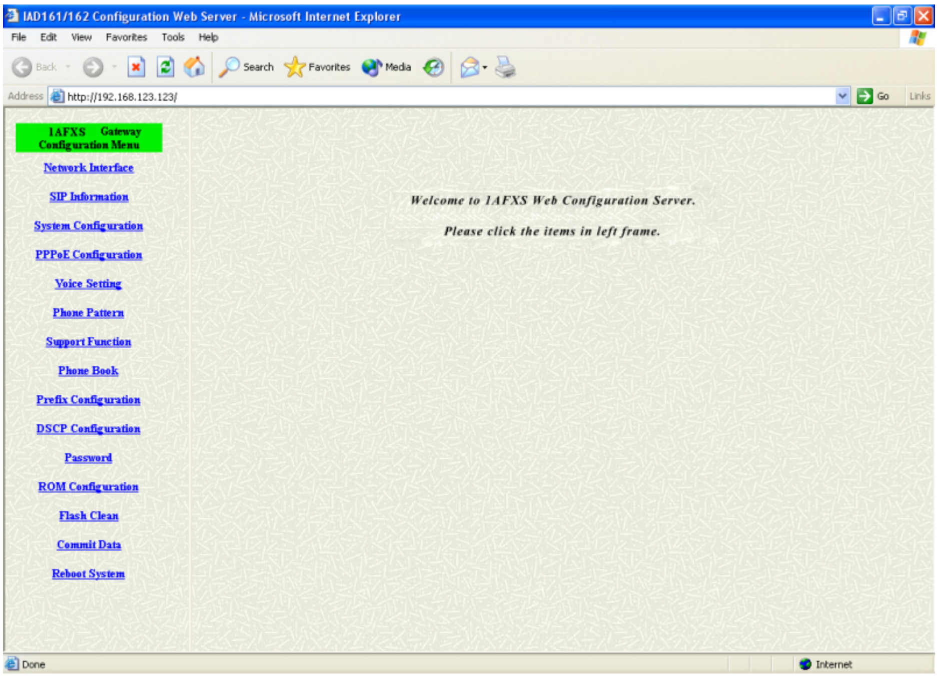


SP5001A/S provides DHCP server function, the Dynamic Host Configuration Protocol is a communications protocol that lets automate the assignment of Internet Protocol (IP) addresses in an organization's network.

9. Connect the SP5001A/S by typing http://192.168.123.123 in the address field of Internet Explorer or Netscape Navigator.

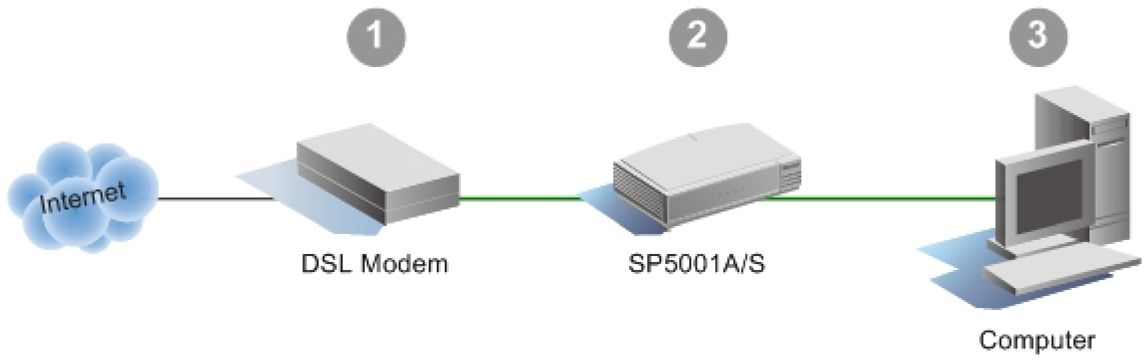


10. Login your gateway by enter **root** as user name and no password when prompted.



If network failed to access the Internet, restart your network in the correct sequence. Failure to restart your network in the correct sequence could prevent you from connecting to the Internet.

1. First, plug in and turn on the broadband modem and wait 1 or 2 minutes.
2. Second, plug in the power to your VoIP gateway and wait 1 minute.
3. Last, turn on your computer.



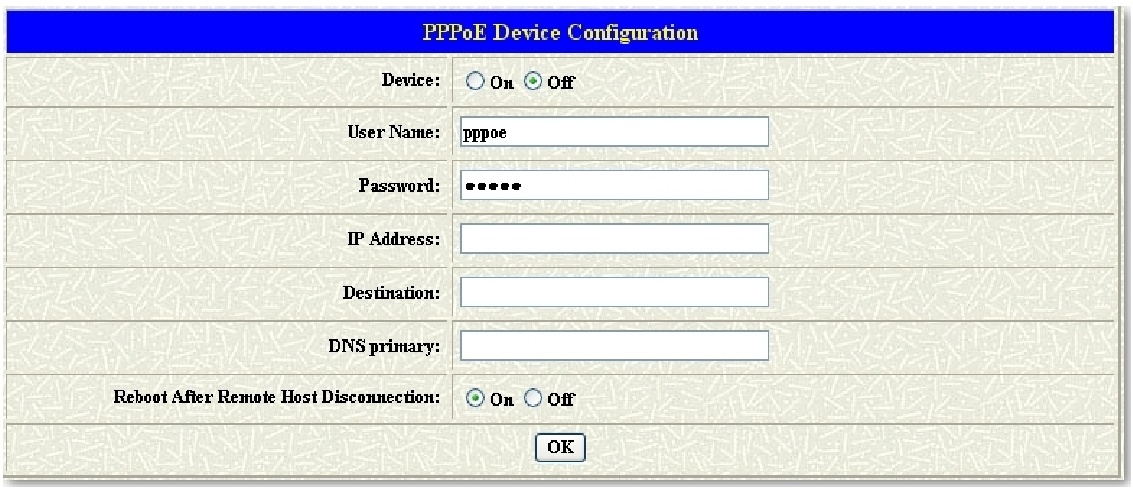
2.2. Internet Connection Setup

This section shows the basic setup to enter the Internet connection settings provided by your ISP. Before proceeding with the Internet connection setup, you need to know the setup information for your specific type of Internet connection, for example, DSL connection or Cable connection, login name / e-mail and password, then you can configure the gateway.

A. PPPoE Connection Setup

For DSL users, many ISPs may require you to log on with a user name (or e-mail address) and password to gain access to the Internet. This connection type is called Point to Point Protocol over Ethernet (PPPoE). PPPoE (Point-to-Point Protocol over Ethernet) is a specification for connecting multiple computer users on an Ethernet local area network to a remote site through common customer premises equipment, which is the telephone company's term for a modem and similar devices, commonly used in dialup connections, users share a Digital Subscriber Line (DSL), cable modem, or wireless connection to the Internet. Most of the PPPoE connection is temporarily assigning an IP address to a requesting Dynamic Host Configuration Protocol (DHCP) NAT router or computer from a pool of IP addresses. The temporary IP address is called a **dynamic IP address**.

1. Select the **[PPPoE Configuration]**



2. Select **[On]** to enable the PPPoE Device

3. Enter your DSL login name into User Name field

4. Enter your DSL password into Password field

5. Click OK button

6. Select **[Commit Data]** and click COMMIT button.

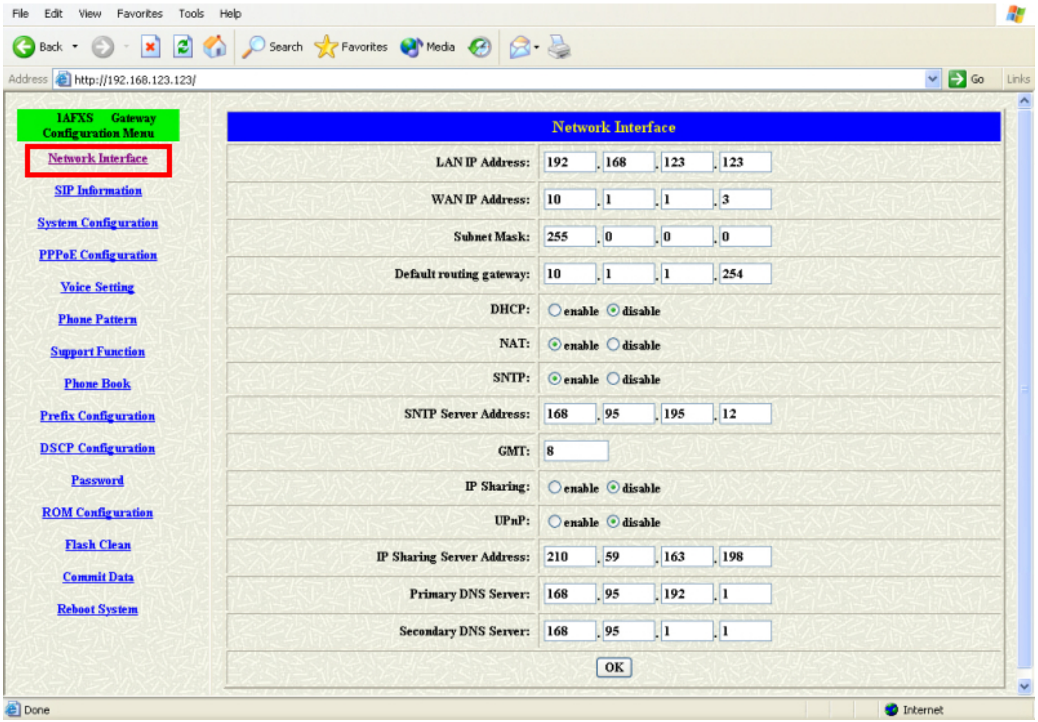
7. Select **[Reboot System]** and click REBOOT button.

Wait for the gateway to reboot, check the DSL connection status by select the **[PPPoE Configuration]**

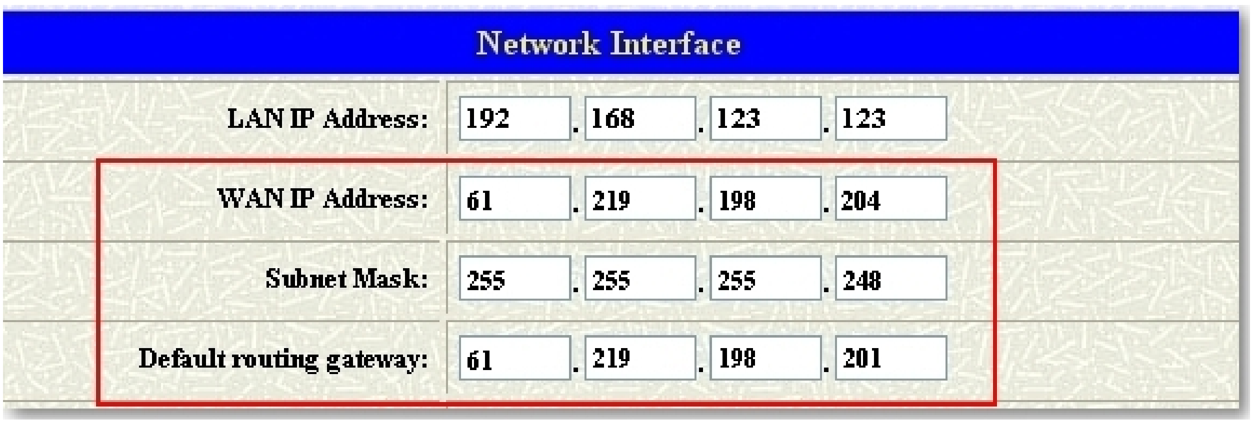
B. Static DSL Connection Setup

A static IP address is a number (in the form of a dotted quad) that is assigned by an Internet service provider (ISP) to be its permanent address on the Internet. VoIP gateways use IP addresses to locate and talk to each other on the Internet, much the same way people use phone numbers to locate and talk to one another on the telephone.

1. Select the **[Network Interface]**



2. Enter the IP address, Subnet and Default Gateway



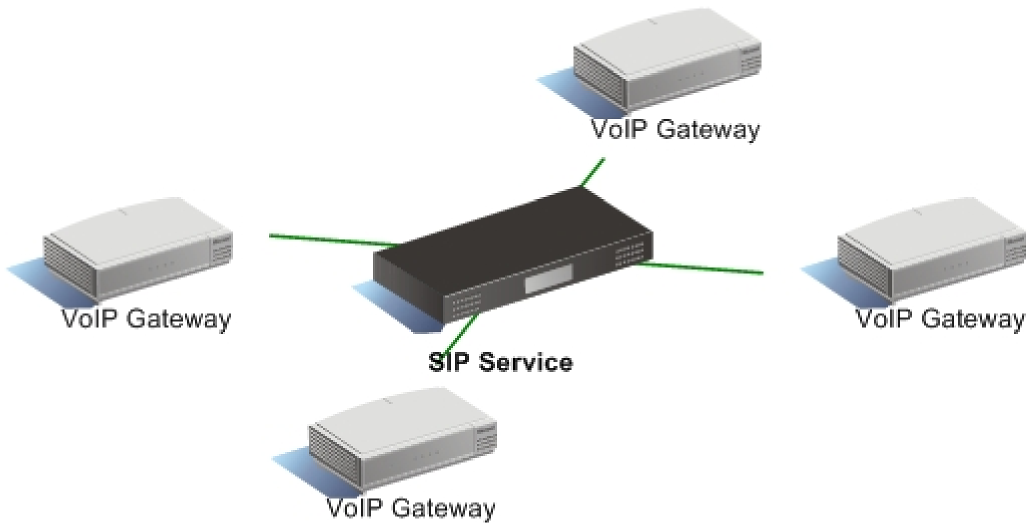
3. Click OK button

4. Select **[Commit Data]** and click COMMIT button.

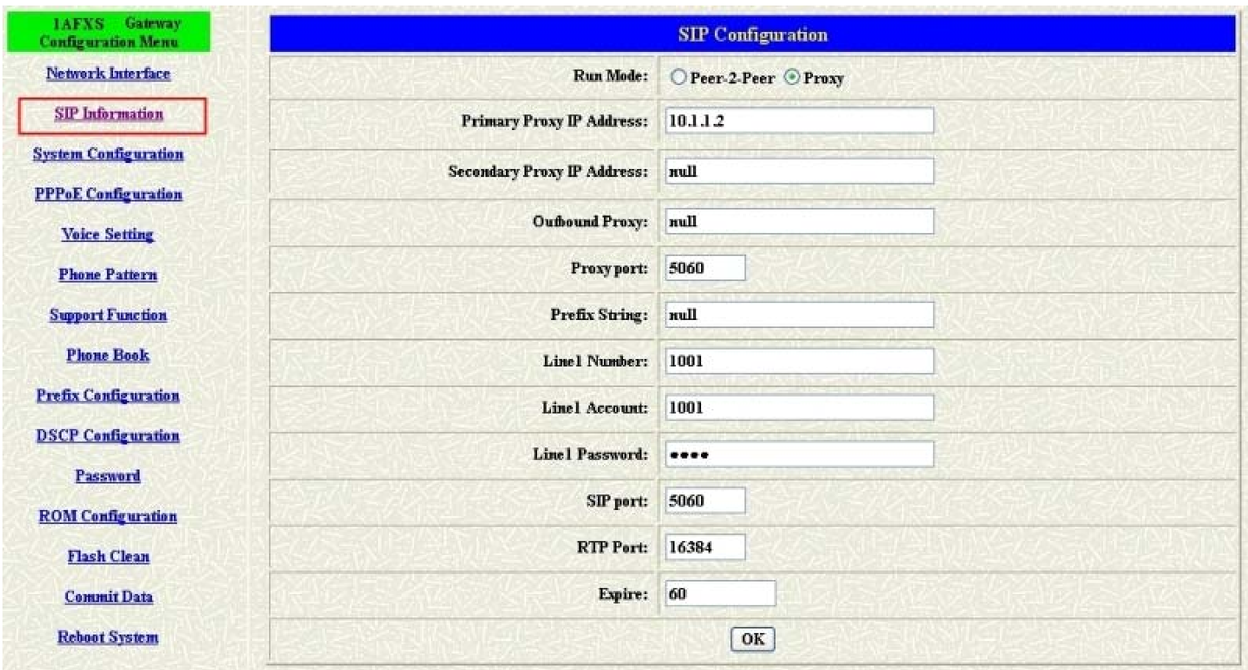
5. Select **[Reboot System]** and click REBOOT button.

2.3. Proxy Mode Setup

You can choose either Proxy mode or Peer-to-Peer mode for communication. Proxy mode requires account information to access to the service; it's assigned by the SIP service provider. SIP Serve rendezvous point at which callees are globally reachable, and perform registration, call routing function. The VoIP gateway (or IP phone) of the users in the domain register their IP addresses with the server so that the other users can reach them. Proxy Mode also suit for the gateway has dynamic IP address connection.



1. Select the **[SIP Information]** at the Configuration Menu section

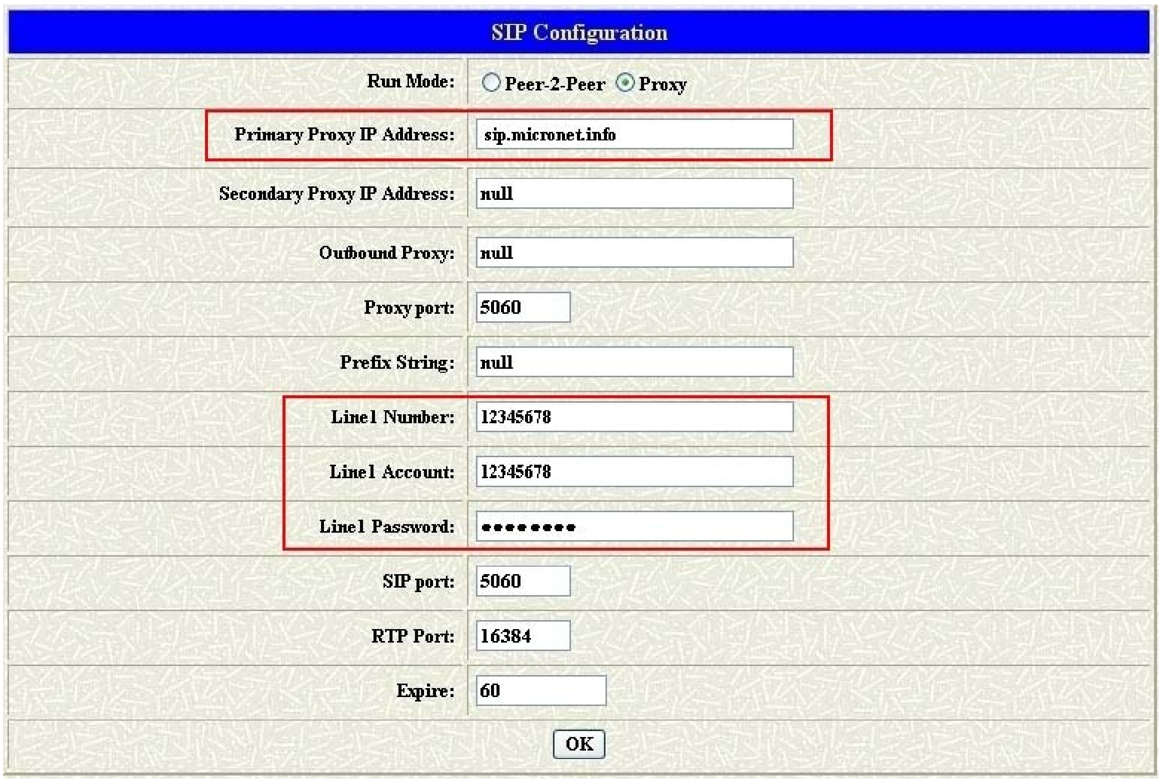


2. Enter the Proxy Server's IP address or URL. For example, **220.130.173.70** or **sip.micronet.info**

3. Enter the Line Number

4. Enter the Account. It can be same as the Line number, the user name or the e-mail account. Check with your VoIP service support for the details.

5. Enter the Password



6. Click OK button

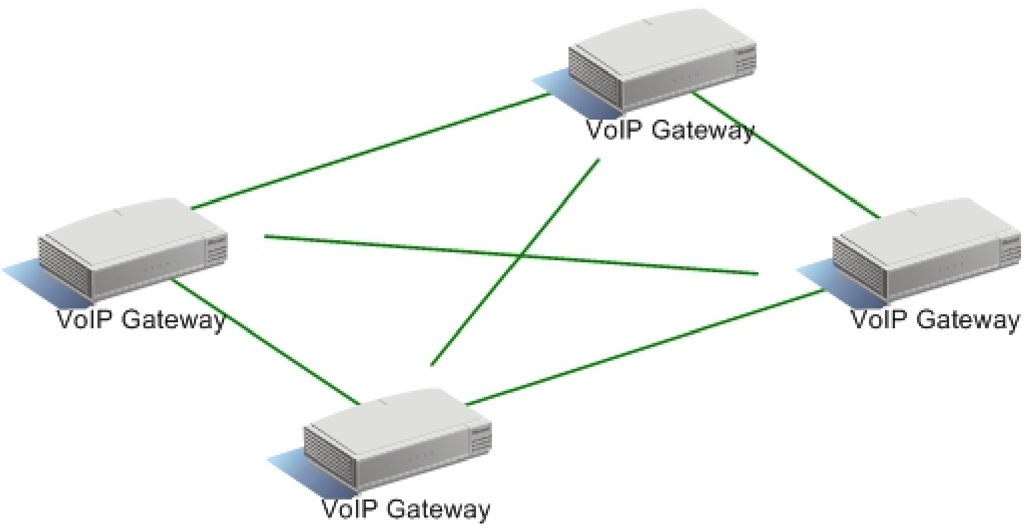
7. Select **[Commit Data]** and click COMMIT button.

8. Select **[Reboot System]** and click REBOOT button.

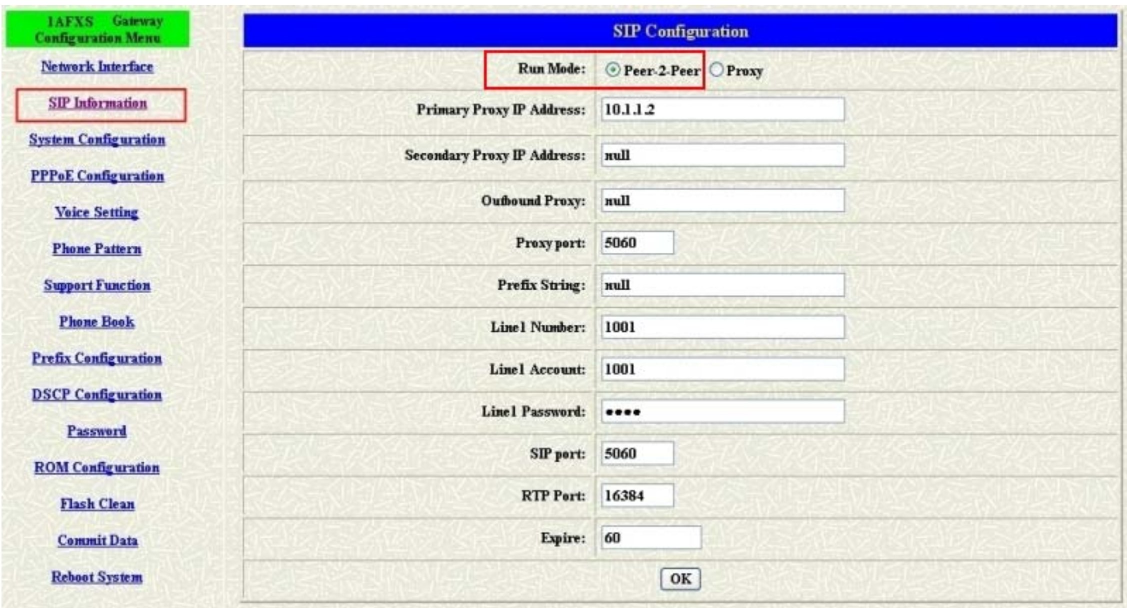
After reboot the SP5001A/S, check the **Status** LED, it shows the gateway has registered to the SIP server successfully when the LED stays on. If not, the **Status** LED is blinking, check the Internet connection and SIP Configuration settings again.

2.4. Peer-to-Peer Mode Setup

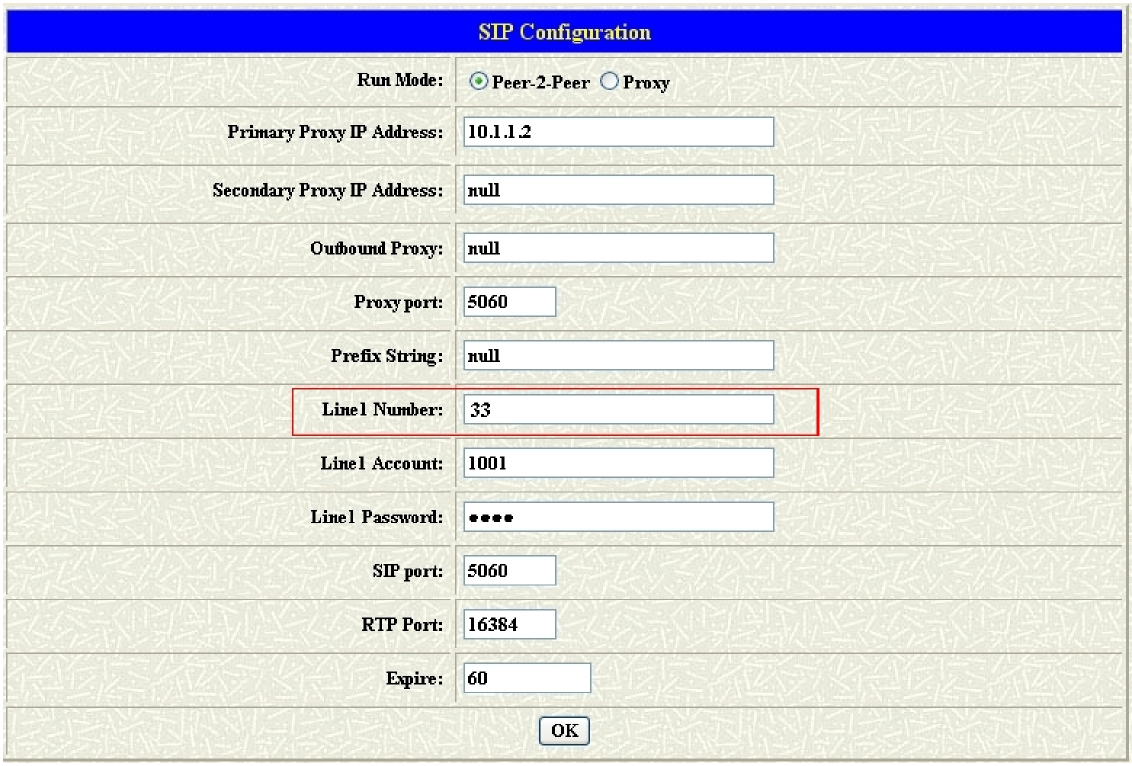
P2P Mode doesn't require any centralized control units like Proxy Mode does, it makes communication between two end-points directly, **[Phone Book]** needs to configure to work with in P2P Mode. It requires direct public IP access, it can also perform the job behind the NAT device with static public IP connection, but it can not work behind the NAT device with the dynamic IP connection.



1. Select the **[SIP Information]** at the Configuration Menu section



2. Enter the Line Number



The Line Number is same as an extension number from of PABX system. You can create your own extension numbering plan for your VoIP system.

3. Click OK button

4. Select the **[Phone Book]** at the Configuration Menu section

5. Enter the destination information into **New Record** field

6. Select **[Commit Data]** and click  button.

7. Select **[Reboot System]** and click  button.

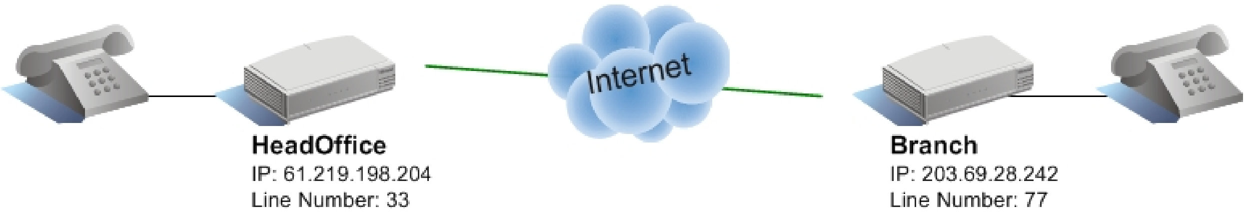
Follow the same steps to configure the remote side gateway. The next section shows the Peer-to-Peer configuration example.

Note:

Remember, the P2P Mode can not work behind the NAT device with the dynamic IP connection.

2.5. P2P Connection Example

The following example shows the gateway's settings of each location in P2P Mode



Head office dials **77#** to reach branch office gateway
Branch office dials **33#** to reach head office gateway

Head Office

Network

WAN IP Address:	61	219	198	204
Subnet Mask:	255	255	255	248
Default routing gateway:	61	219	198	201

SIP Line Number

Line1 Number:	33

Phone Book

Phone Book				
Index	Name	IP_Address	e164	Port
1	Branch	203.69.28.242	77	

Branch Office

Network

WAN IP Address:	203	69	28	242
Subnet Mask:	255	255	255	0
Default routing gateway:	203	69	28	254

SIP Line Number

Line1 Number:	77

Phone Book

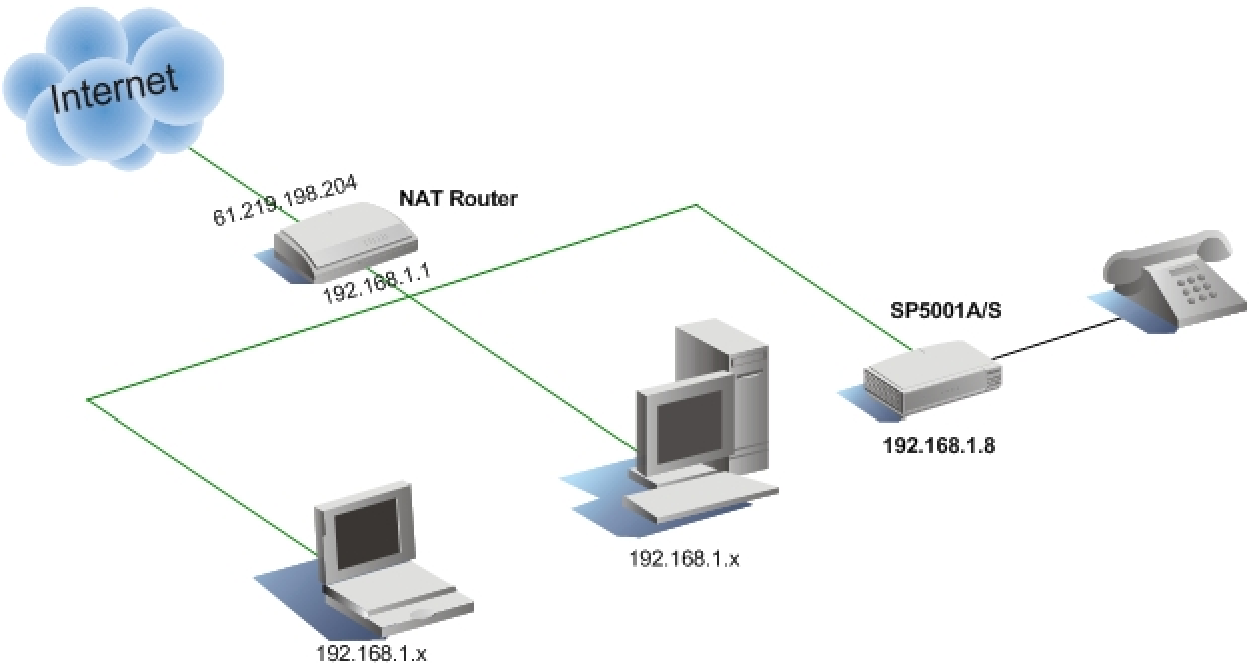
Phone Book				
Index	Name	IP_Address	e164	Port
1	HeadOffice	61.219.198.204	33	

3. Advanced Setup

It would be simple if every VoIP gateway, or computer that connects to the Internet could have its own static IP number, but when the Internet was first conceived, the architects didn't foresee the need for an unlimited number of IP addresses. Consequently, there are not enough IP numbers to go around and we use the NAT device or router connects our local area network (LAN), or the group of PCs in your home or office, to the Internet. In this section, we will show you how to configure your SP5001A/S behind the NAT device if your SP5001A/S acts standalone device in your network.

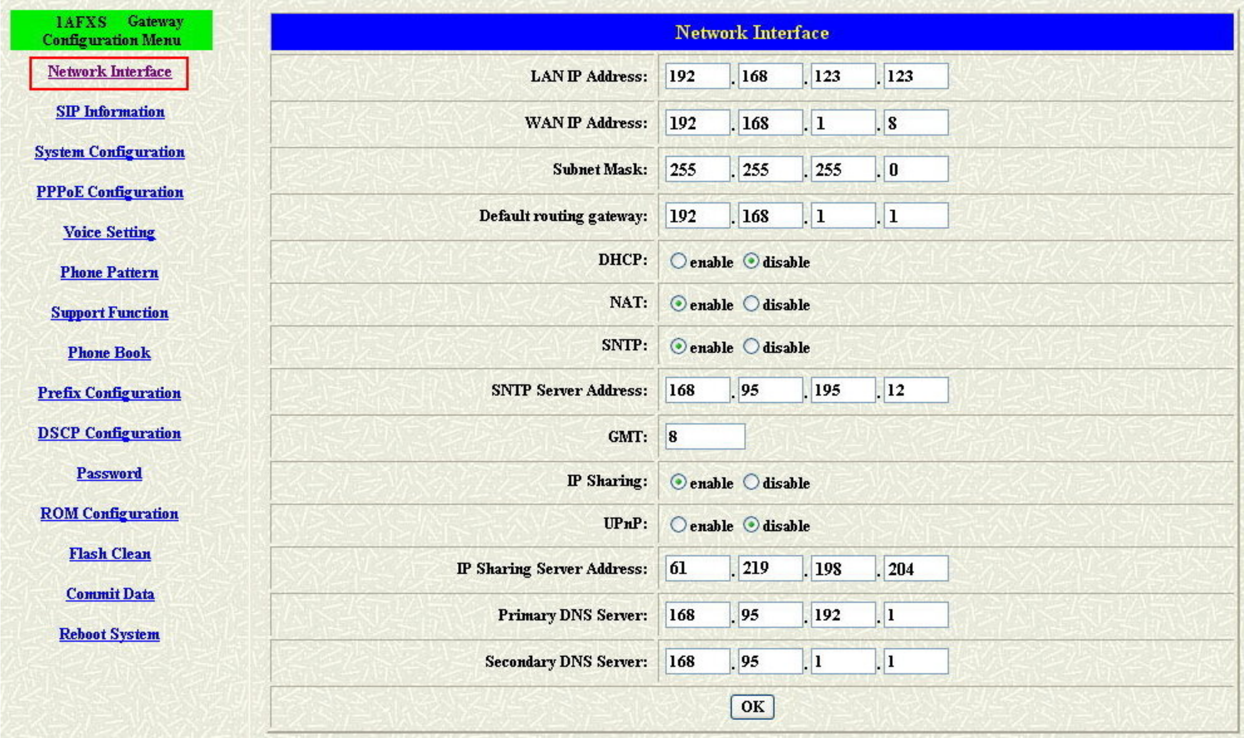
3.1. Behinds the NAT Router (P2P Mode)

When you place the SP5001A/S behinds the NAT router in P2P Mode, a few more settings need to configure on the SP5001A/S and router.



Setup the SP5001A/S:

1. Open the web browser to connect the gateway and select the **[Network Interface]**



2. Change the WAN IP Address, assign the IP address depends on your router settings, for example: **192.168.1.8**

   Note: The static IP must configured in this application

3. Change the Subnet Mask if necessary, for example: **255.255.255.0**

4. Change the WAN IP Address. Here means your router's LAN IP address, for example: **192.168.1.1**

5. Select the **[IP Sharing]** and enable

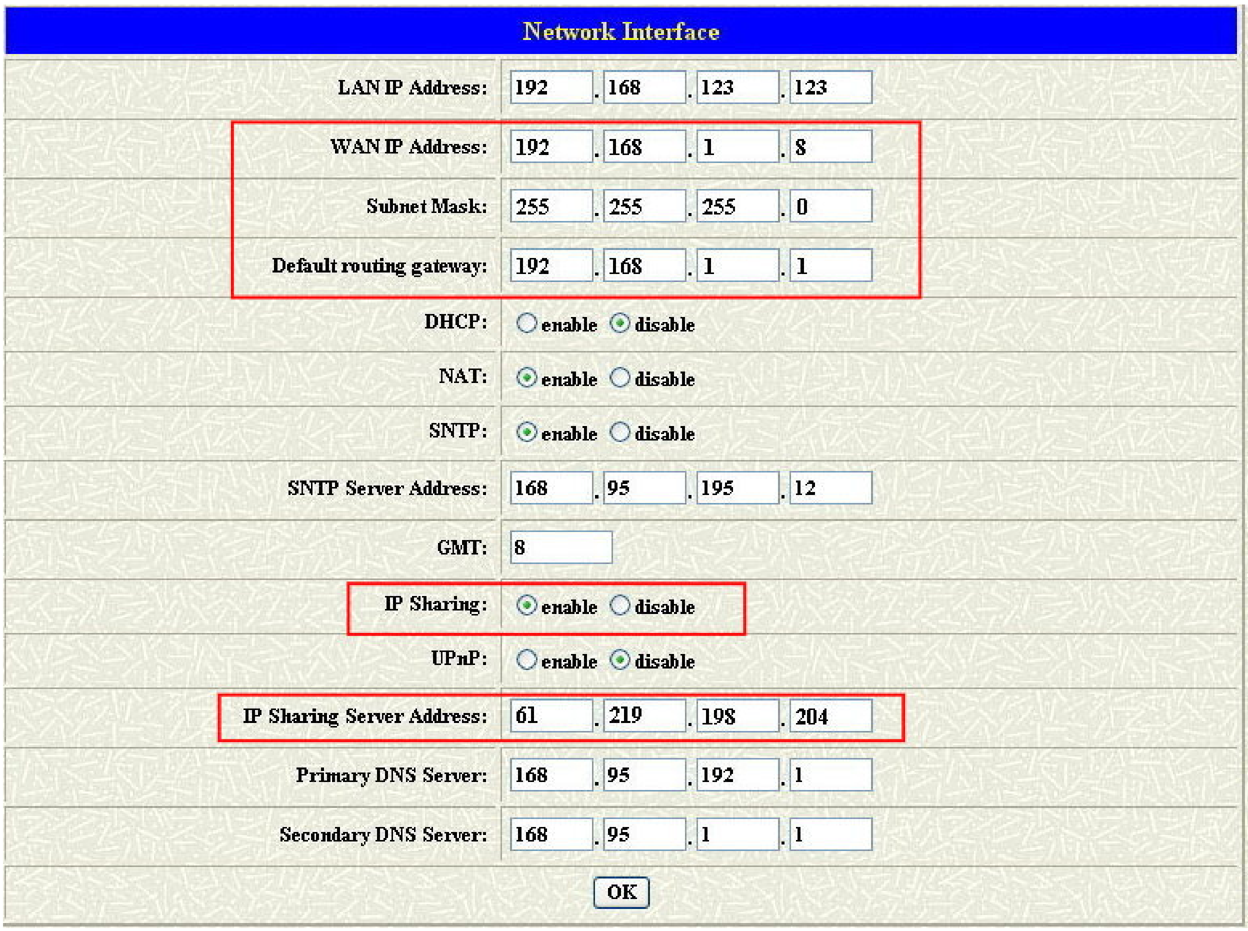
6. Enter the **IP Sharing Server Address**, here means your router's WAN IP address, for example: **61.219.198.204**

7. Click the 

button

8. Select **[Commit Data]** and click COMMIT button.

9. Select **[Reboot System]** and click OK button.



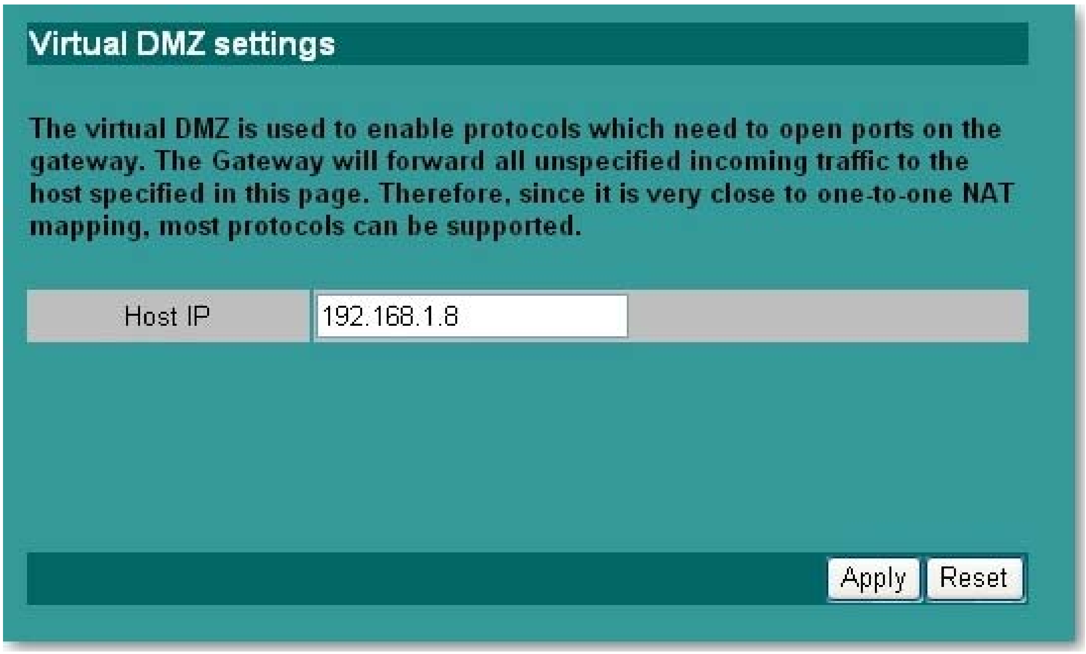
Setup the NAT router:

When the VoIP gateway or computer behind the NAT router, normal web surfing and email will not know the difference, but some communication services using ports other then the normal web ports (ports are like door ways to your computer and for security the communication has the abnormal ones closed and locked but the normal ones open like the web surfing port 80). It must enable the **DMZ** (Demilitarized Zone) function or setup the **Port Forwarding**(or called Virtual Server), let the communication traffic can pass through the router.

A. DMZ Setup

1. Enter the NAT router configuration by web browser or software utility.

2. Locate the DMZ function and enable it



SP888B Broadband Router Screen Shot

DMZ

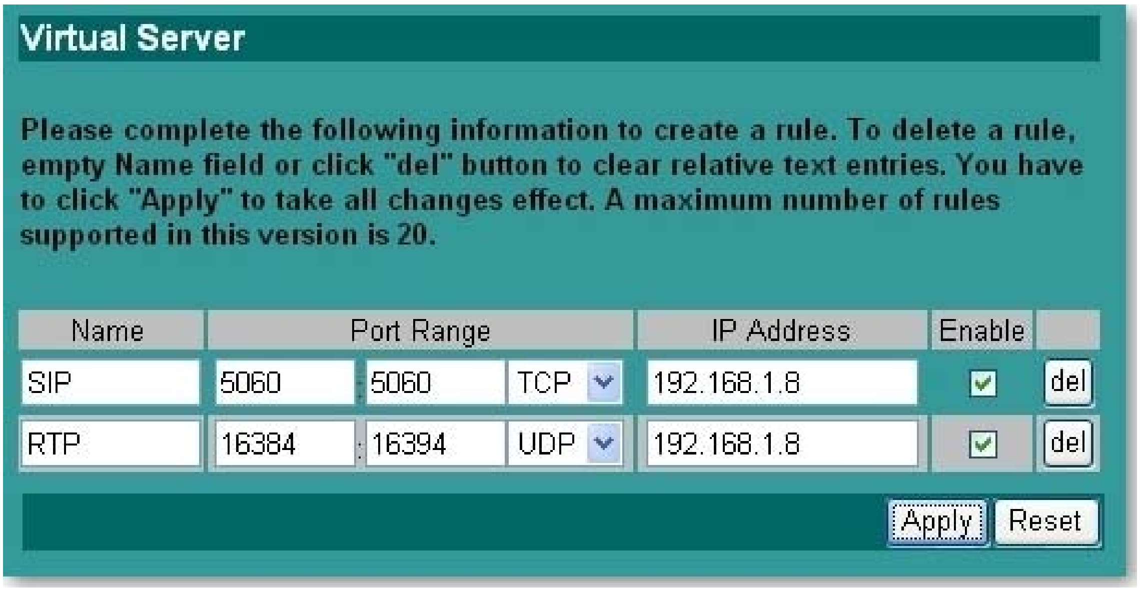
In computer networks, a DMZ (demilitarized zone) is a computer host or small network inserted as a "neutral zone" between a company's private network and the outside public network. It prevents outside users from getting direct access to a server that has company data. (The term comes from the geographic buffer zone that was set up between North Korea and South Korea following the UN "police action" in the early 1950s.)

B. Port Forwarding (Virtual Server)

If you can not find the DMZ function on your router or firewall, then the Port Forwarding is another way to allow the communication traffic pass through. A broadband router creates a firewall between your internal network and the internet. A firewall keeps unwanted traffic from the internet away from your LAN computers. A ‘tunnel’ can be created through your firewall so that the computers on the Internet can communicate to one of the computers on your LAN on a single port. This is handy for running web servers, game servers, ftp servers, VoIP applications or even video conferencing. This is called port forwarding. Port 5060 and Port 16384 ~ are commonly used in SIP from Micronet VoIP products

1. Enter the NAT router configuration by web browser or software utility.

2. Locate the Port Forwarding function and enable it



SP888B Broadband Router Screen Shot

Port

Applications running on TCP/IP open connections to other computers using something called ports. Ports allow multiple applications to reside on a single computer - all talking TCP/IP. Ports are another set of numbers AFTER the standard IP address. Applications often hide these port numbers to reduce the complexity of TCP/IP. Example: web services (HTTP) reside on port 80 by default, port 5060 is for the SIP signaling by default, port 16384 ~ is for the RTP by default, etc.

3.2. Codec Selection

Codec (Coder / Decoder)

Codecs are used to convert analog signals to a digital bit stream, and another identical codec at the far end of the communication converts digital bit stream back into an analog signal. Codecs vary in the sound quality, the bandwidth required, the computational requirements, etc. Codecs generally provide a compression capability to save network bandwidth. Some codecs also support silence suppression, where silence is not encoded or transmitted. In the VoIP world, codec's are used to encode voice for transmission across IP networks.

Micronet VoIP gateway supports several different codecs, G.711A/μ law, G723.1, G729, and when talking to each other, negotiate which codec they will use.

Codec	Description	Bit Rate (Kb/s)	Remark
G.723.1	G.723.1 is an ITU-T standard codec. Its reasonably low bit rate (6.3Kbps or 5.3Kbps). Use of this codec in a product requires licensing by Sipro Lab Telecom	5.6 / 6.3	It encodes speech or other audio signals in frames using linear predictive analysis-by-synthesis coding.
G.729	Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear-prediction (CS-ACELP)	8	Low delay (15 ms)
G.711	G.711 is the international standard for encoding telephone audio on an 64 kbps channel. It is a pulse code modulation (PCM). This is most	64	mu-law (US, Japan) and A-law (Europe) companding

All VoIP packets are made up of two components: **voice samples** and **IP/UDP/RTP headers**. Although the voice samples are compressed by the Digital Signal Processor (DSP) and may vary in size based on the codec used, these headers are a constant 40 bytes in length. This table shows the nominal Ethernet bandwidth consumption.

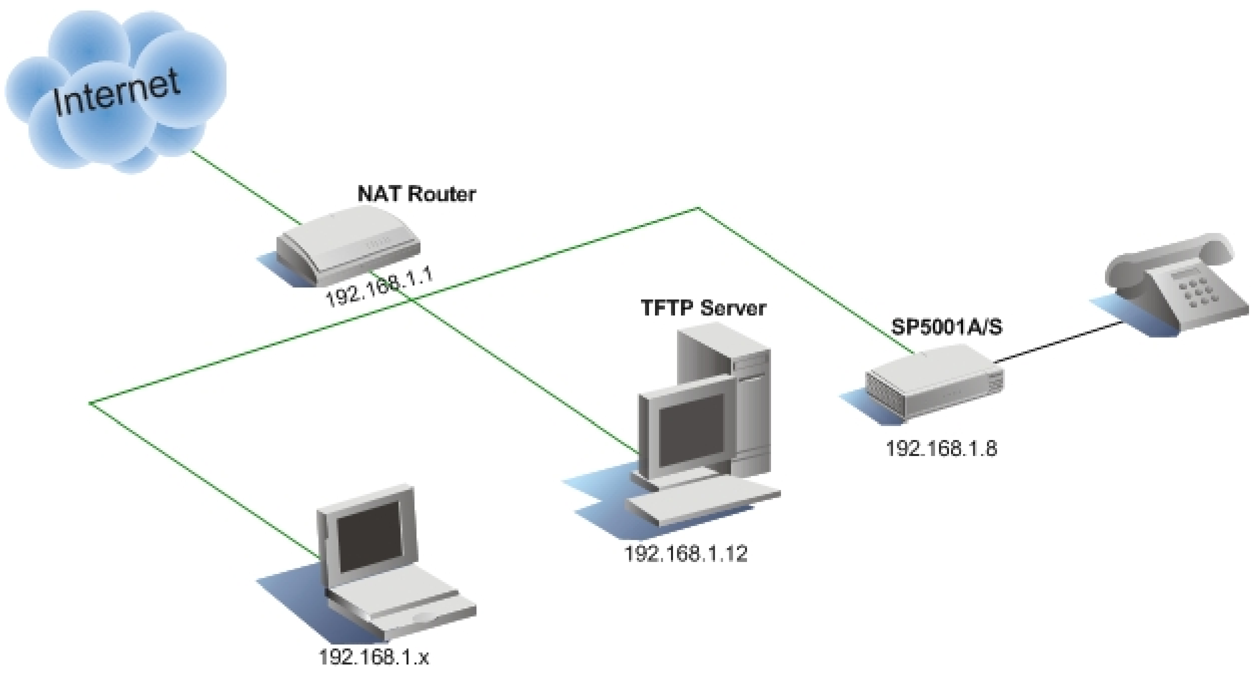
Codec	Bit Rate (Kb/s)	Nominal Ethernet Bandwidth
G.723.1	5.3 / 6.4	20.8 kbps (for 5.3 frame bit rate) 21.9 kbps (for 6.4 frame bit rate)
G.729	8	31.2 kbps
G.711	64	87.2 kbps

4. Firmware Upgrade

Firmware is a combination of software and hardware. Computer chips that have data or programs recorded on them are firmware. These chips commonly include the following: **ROMs** (read-only memory), **PROMs** (programmable read-only memory), **EPROMs** (erasable programmable read-only memory), it's same as software, except it is executed from ROM, and does not disappear when the power is turned off. Firmware in PROM or EPROM is designed to be updated if necessary through a software update. You can download firmware updates for Micronet VoIP products from Micronet web site at Download Center.

You must have a TFTP or FTP server configured and running to perform the download operation, you can download the TFTP program from Micronet web site.

Note:
Firmware should be upgraded ONLY if you experience problems with the Gateway

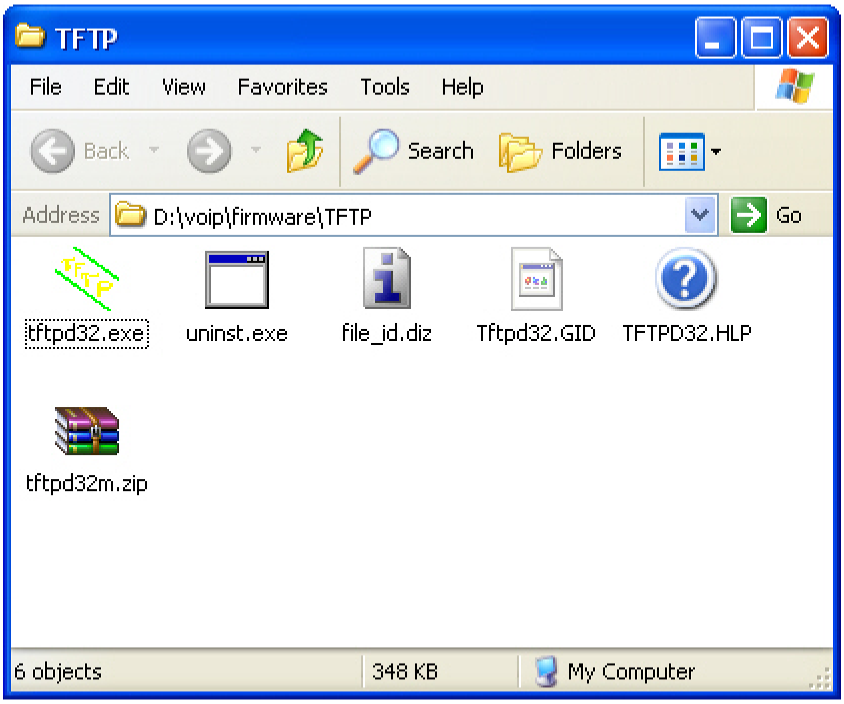


4.1. TFTP Server Setup

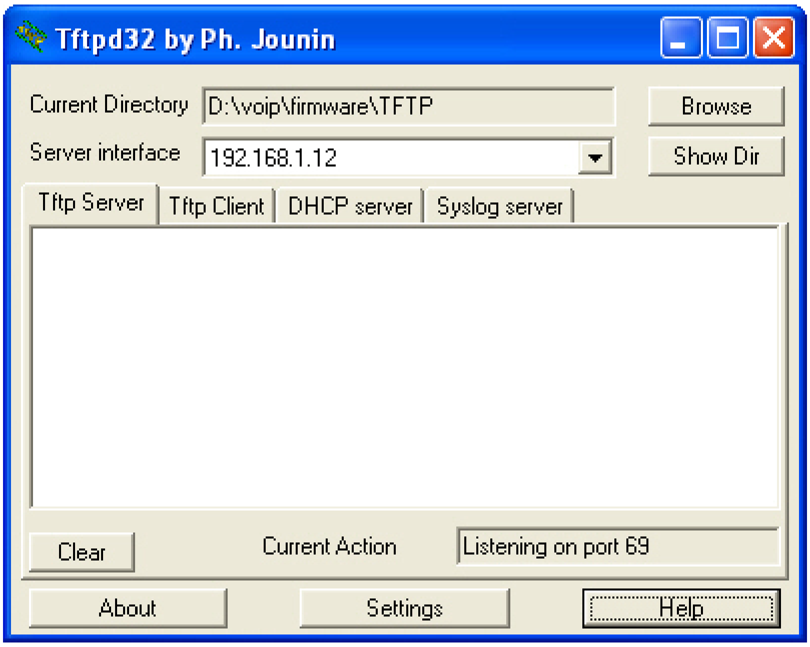
1. Click the link to download the TFTP program. The file downloads as a compressed file called a zip (.zip) file.
   http://www.micronet.info/Download/Driver/VoIP/utility/tftpd32m.zip

   Note:
   The file opens in WinZip® or another decompression program, use the program to extract the zip from the compressed file.

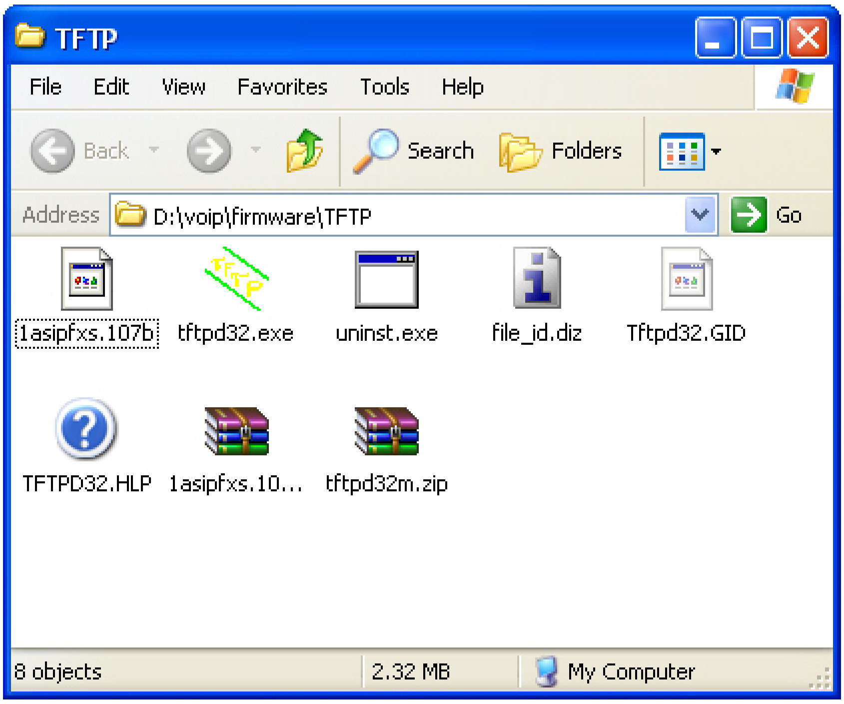
2. Select the folder where you want to save the compressed file, and then click Save



3. After extracting the file, you can double-click the **tftpd32.exe** to start the program.



4. Download the firmware, copy and decompress the file into same folder where the TFTP program located

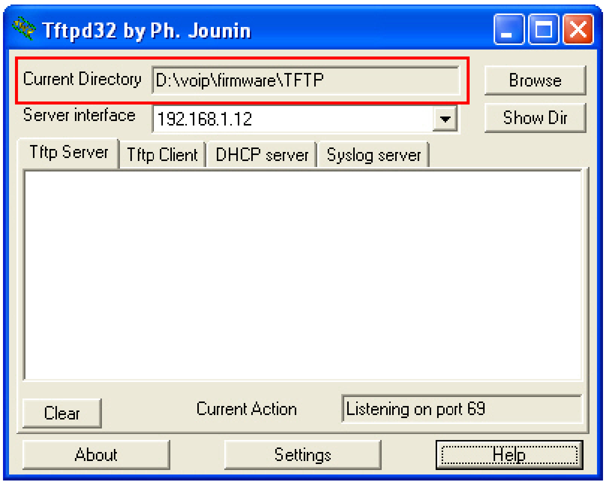


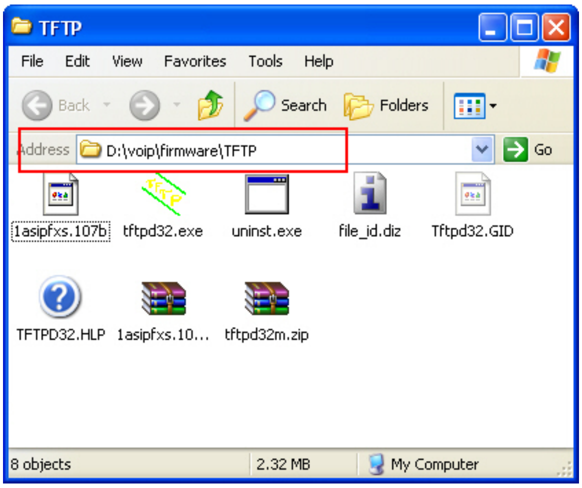
Firmware file name: **1asipfxs.107b**

5. Now, you have the TFTP server and latest Application firmware ready. Go to next section to configure the SP5001A/S.

Note:

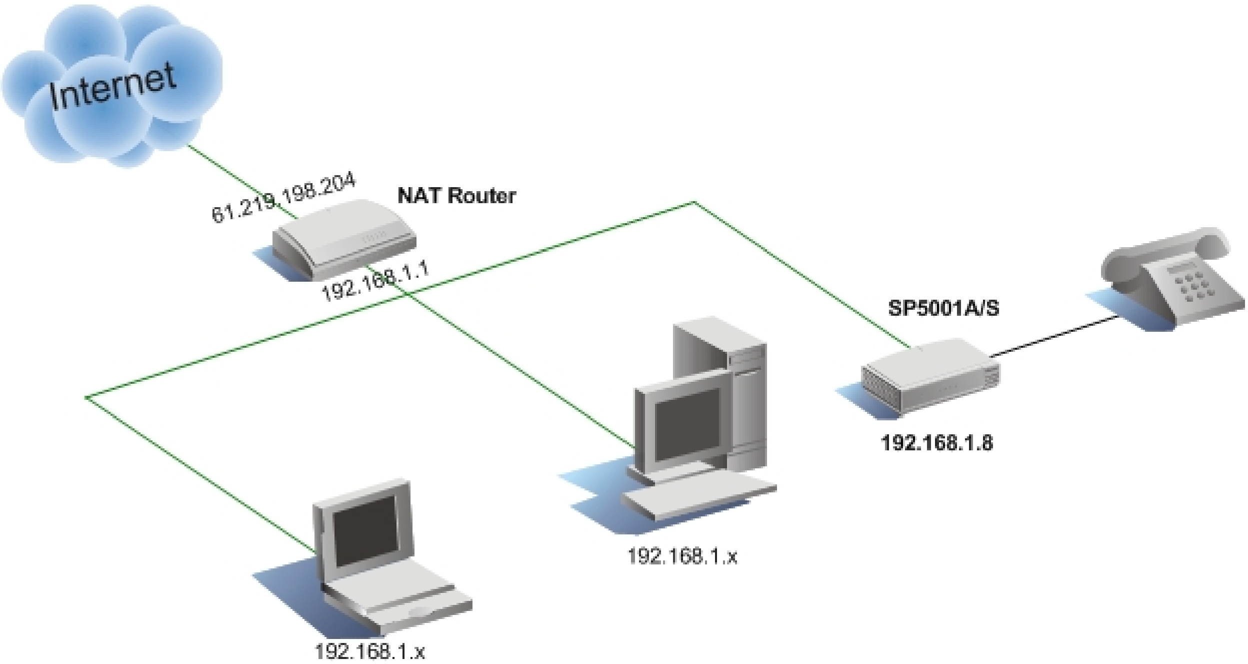
Make sure the **Current Directory** has located the same folder as where the firmware file saved.





4.2. Upgrade by WEB Interface

Assumed your SP5001A/S has configured static IP address (**192.168.1.8**) as the diagram showed.



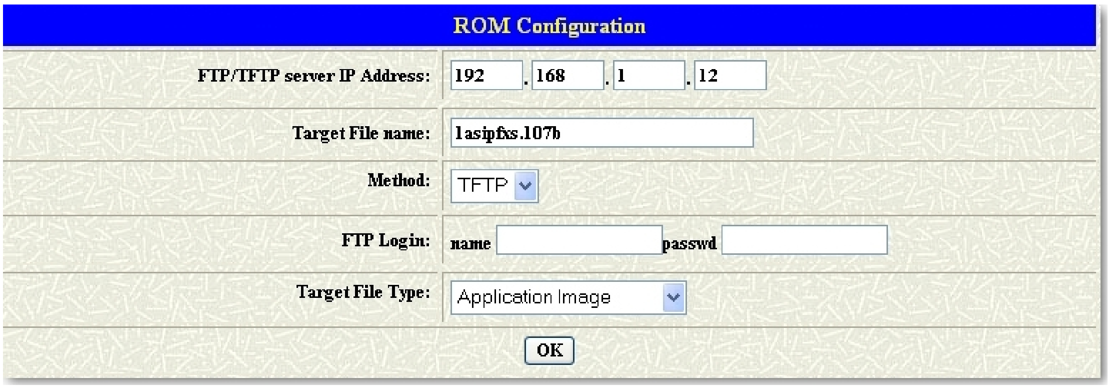
1. Open the web browser to connect the gateway and select the **[ROM Configuration]**

2. Enter the TFTP server IP address (192.168.1.12)

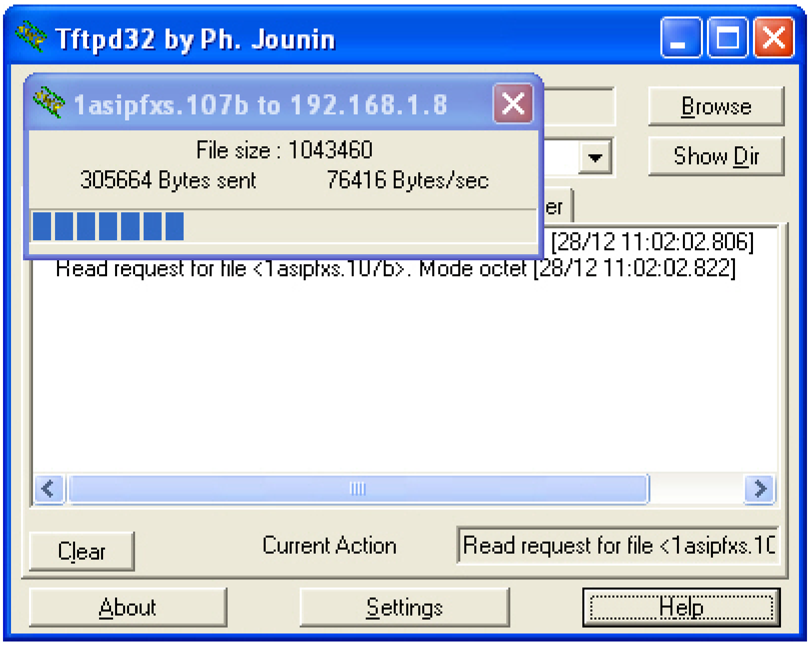
3. Enter the firmware file name into Target File name field

4. Select **[TFTP]** method

5. Select **[Application Image]** as Target File Type



6. Click OK button to start downloading the firmware



TFTP server is uploading the file to the gateway

After file transferred complete, the gateway will write the new firmware into the Flash ROM, wait until see the web browser showed **Please issue FLASH CLEAN to consist software version**

7. Select [Flash Clean] menu and click the 

 button

When the screen showed **Flash cleaned!!** You can now reboot the gateway.

Note:

All the settings will be erased after upgrade the firmware, the gateway needs to re-configure again.

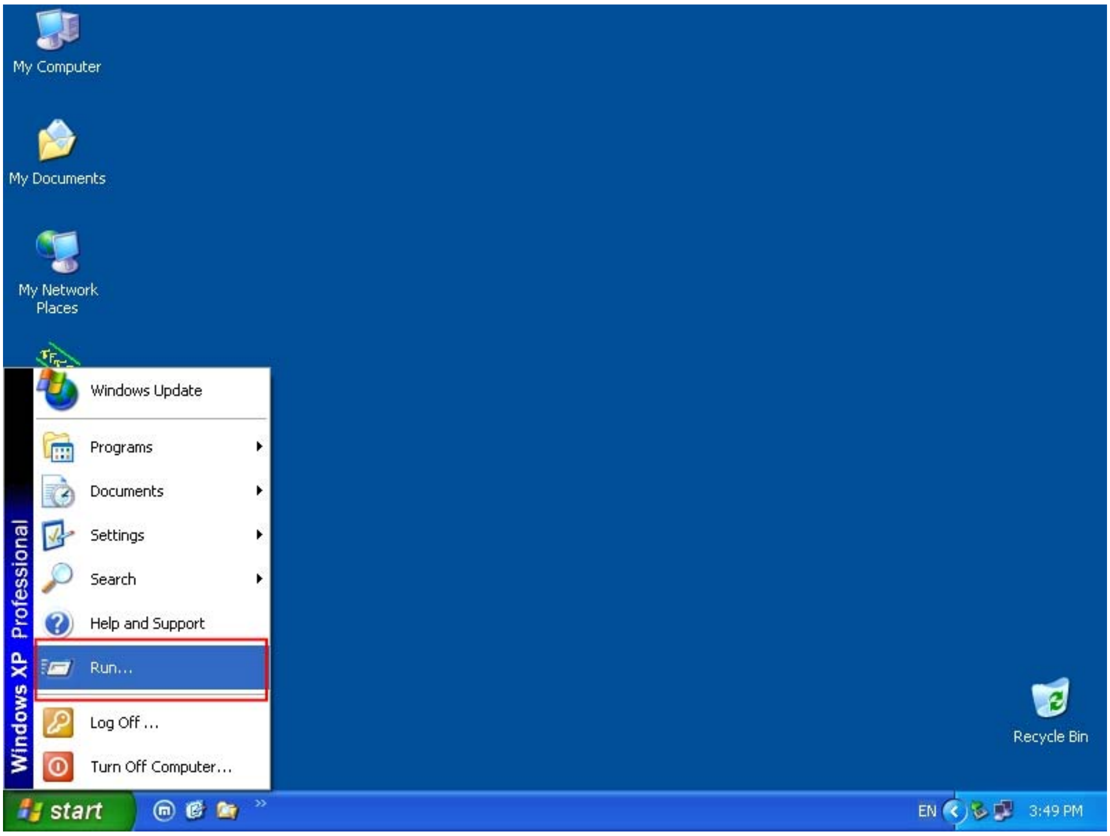
You can change Micronet VoIP gateway's protocol from SIP to H.323, or from H.323 to SIP by firmware uploaded as well.

4.3. Upgrade by Telnet Command

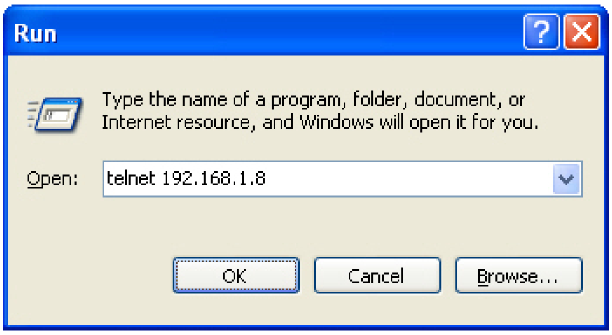
Telnet is another way you can access the gateway, assuming it has given you permission. More technically, Telnet is a user command and an underlying TCP/IP protocol for accessing remote network devices. On the Web, HTTP and FTP protocols allow you to request specific files from remote network devices, but not to actually be logged on as a user of that computer. With Telnet, you log on as a regular user with whatever privileges you may have been granted to the specific application and data on that device. Telnet is most likely to be used by program developers and anyone who has a need to use specific applications or data located at a particular host.

A Telnet command request looks like this: **telnet 192.168.1.8**
We will show you how to use the telnet command to do the upgrade firmware under Windows OS. For the Linux OS users, please check your OS manual or ask the network administrator for the help.

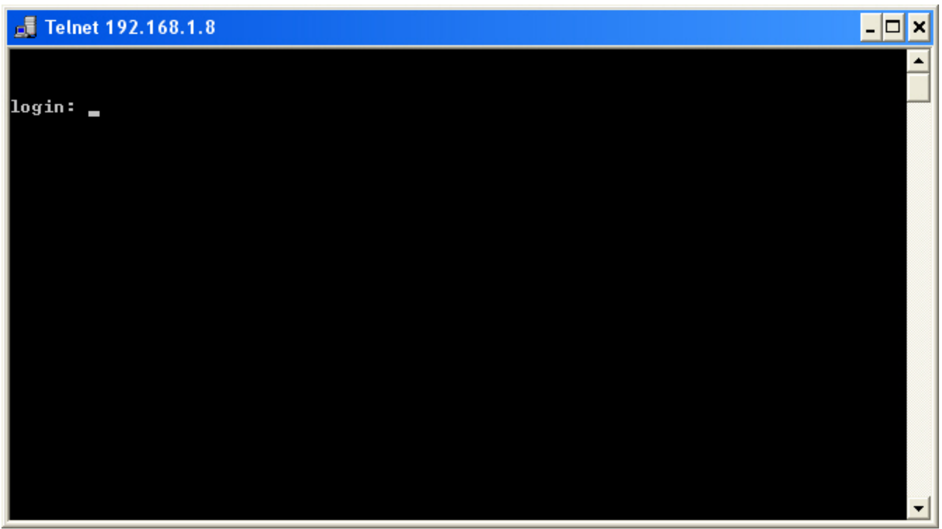
1. Click **Start** and select **Run…**



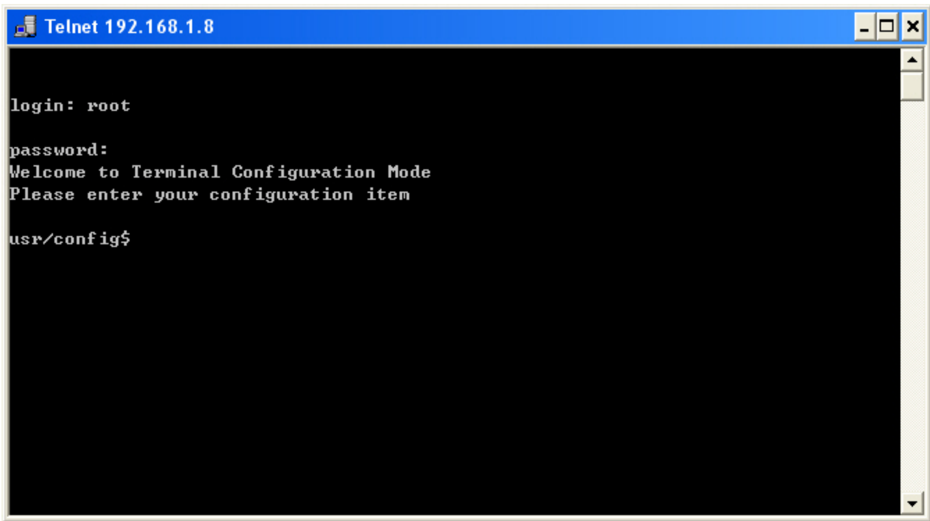
2. Type **telnet 192.168.1.8** and click  to connect your gateway



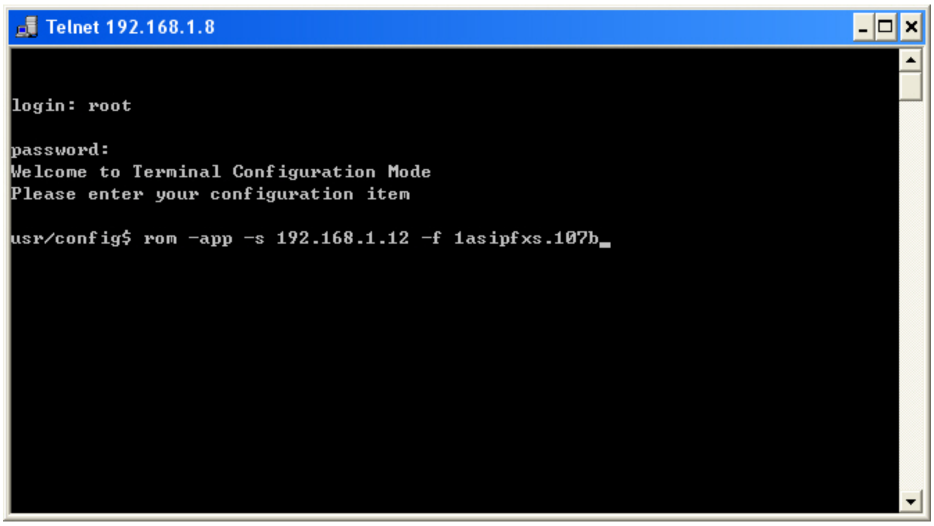
3. Login screen



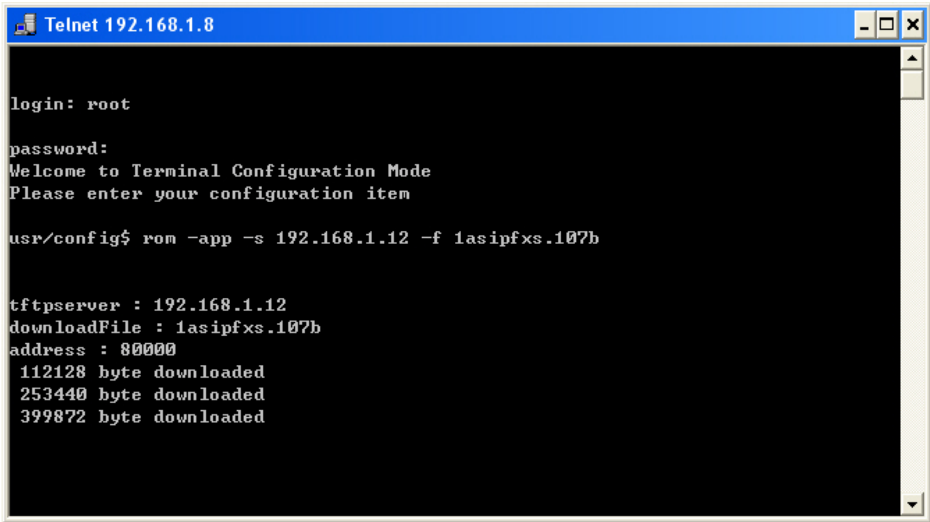
4. Type **root** as login name, no password, press **[Enter]**



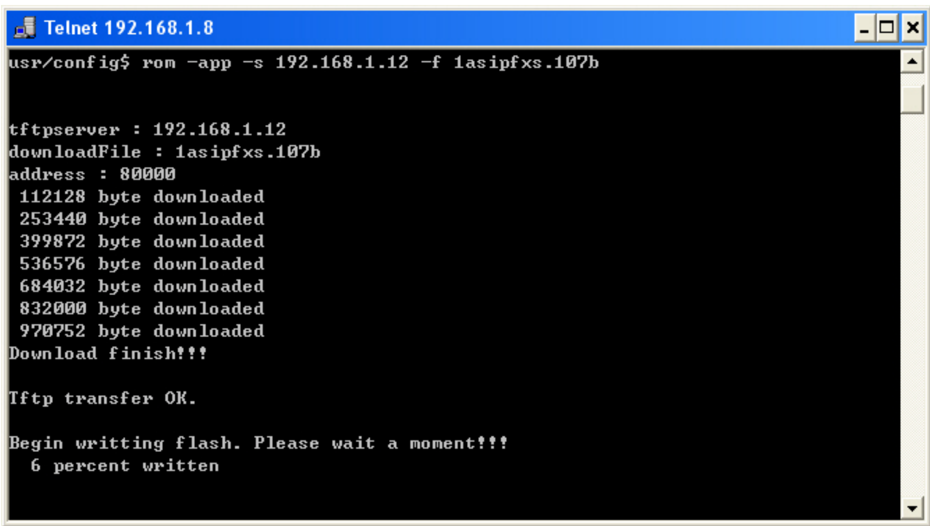
5. Type the following command line to execute the upgrade procedure.
   usr/config$ **rom -app -s 192.168.1.12 -f 1asipfxs.107b**



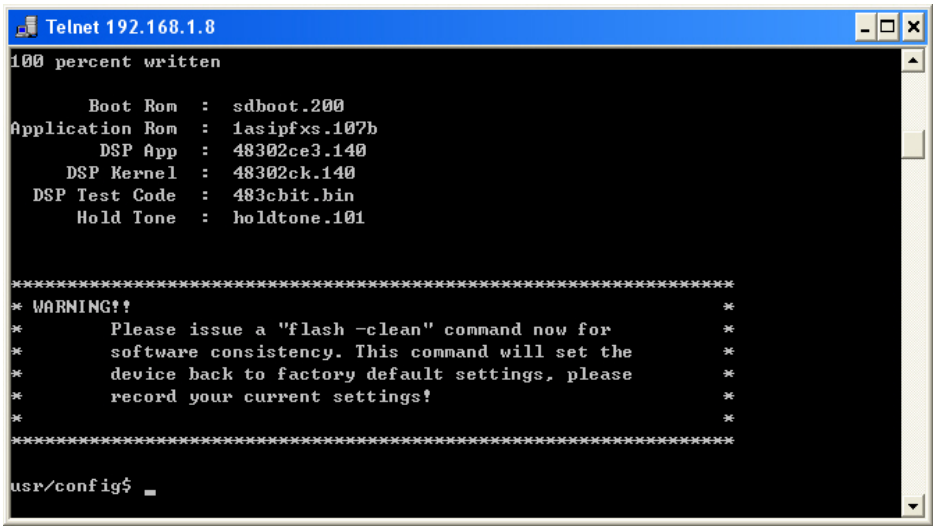
6. Gateway is downloading the firmware file



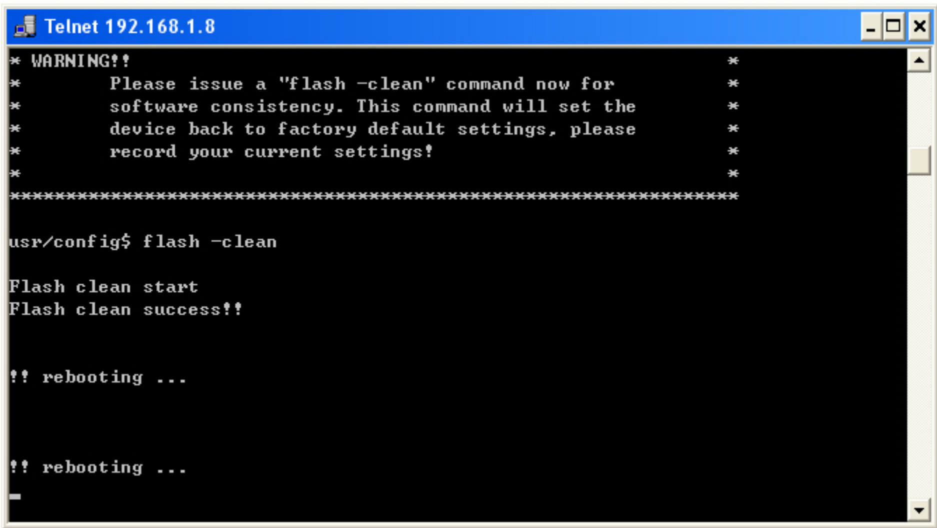
7. After downloaded the file, start writing into Flash ROM



8. Always clean the flash memory after upgraded new firmware.



9. After the **rebooting ...** message showed, close the Telnet windows



10. Re-login the gateway and configure it.

Note:

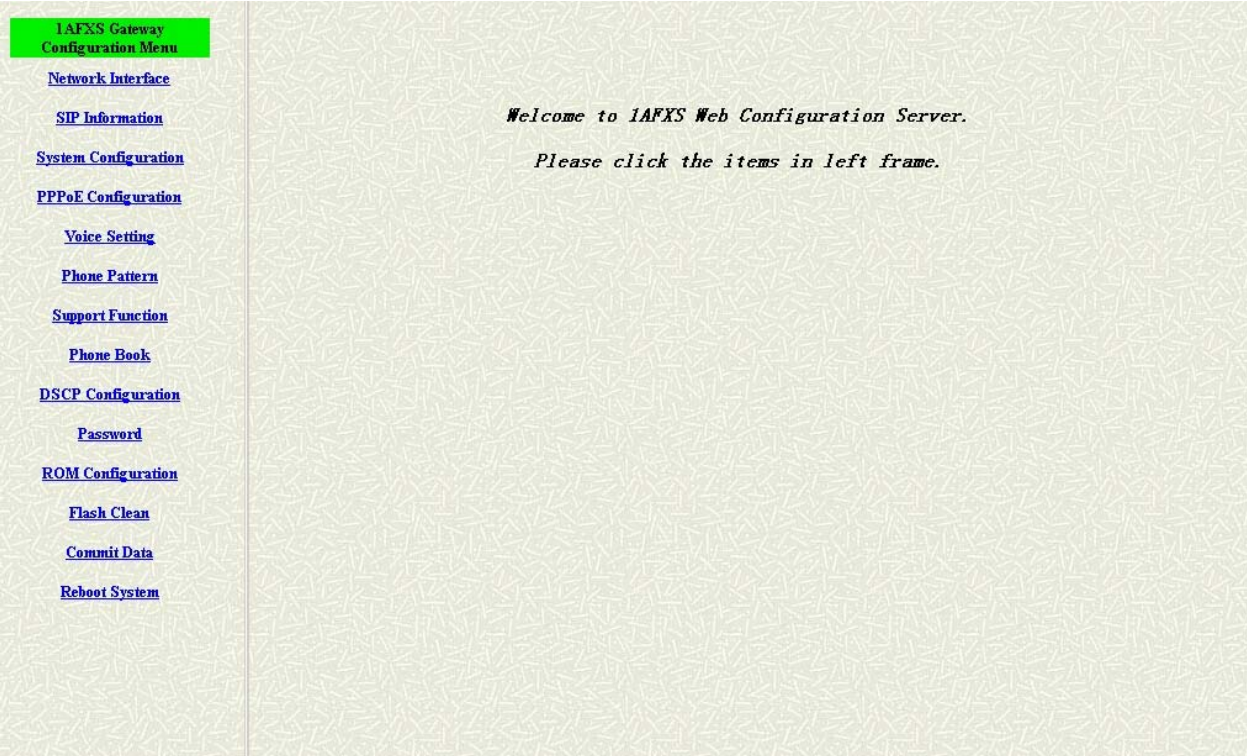
Telnet mode is good for user to monitoring the upgrade procedures

For more details about the Telnet commands, refer to the **Command List** section

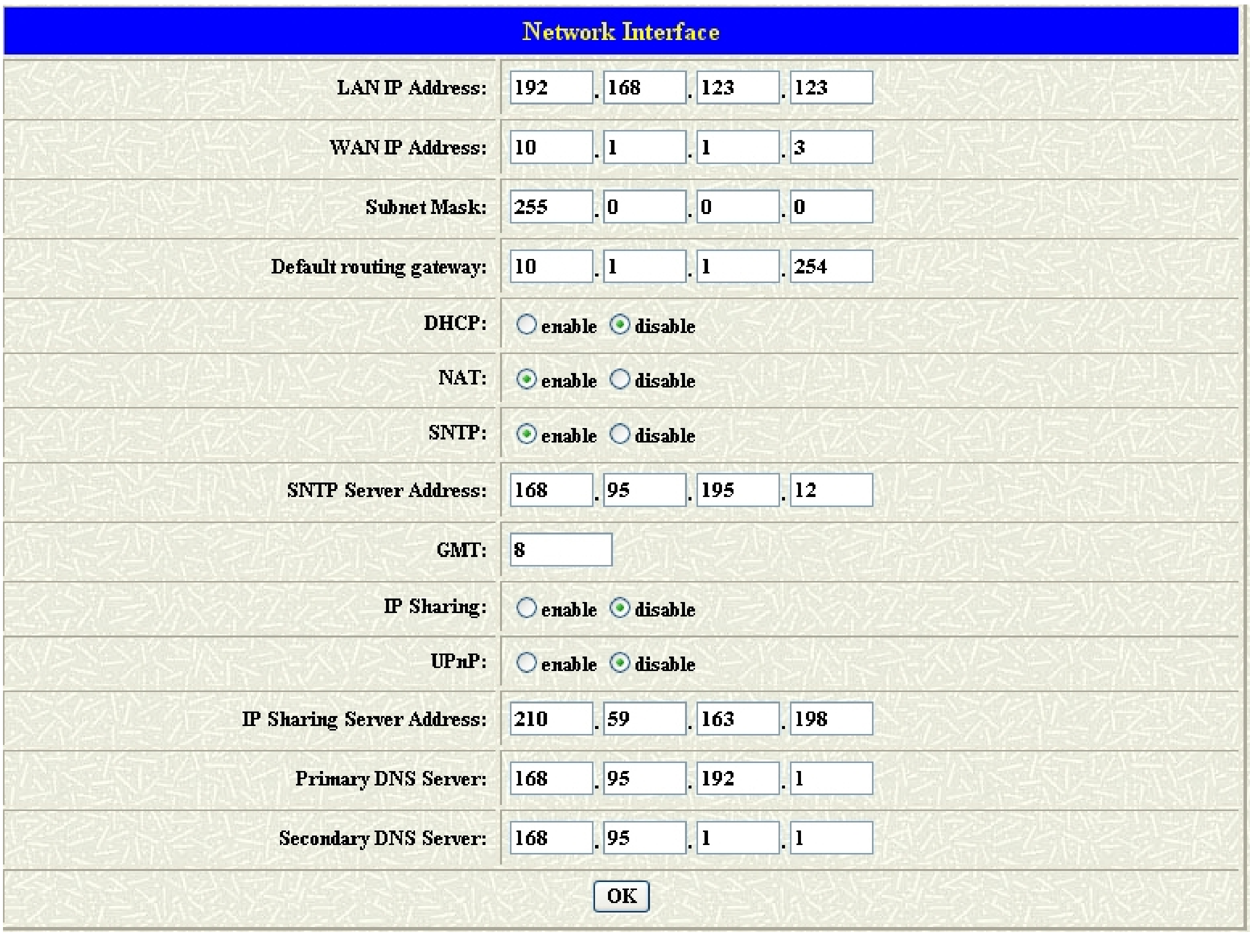
5. WEB Configuration Menu

Micronet gateway provides a built-in web server. You can be accessed via Microsoft Internet Explorer or Netscape Navigator through use of a computer connected with an Ethernet cable to the SP5001A/S gateway. Configuration and administration can be performed through this convenient web interface. This section shows all of the configure functions.

Main page



5.1. Network Interface



LAN IP Address When the gateway connects to the DSL modem directly and shares the connection for the LAN devices, this IP Address is assigned for the LAN devices' default gateway

If the gateway acts stand alone device and behinds the NAT router, the LAN IP Address can be leave as it is and disable the NAT function

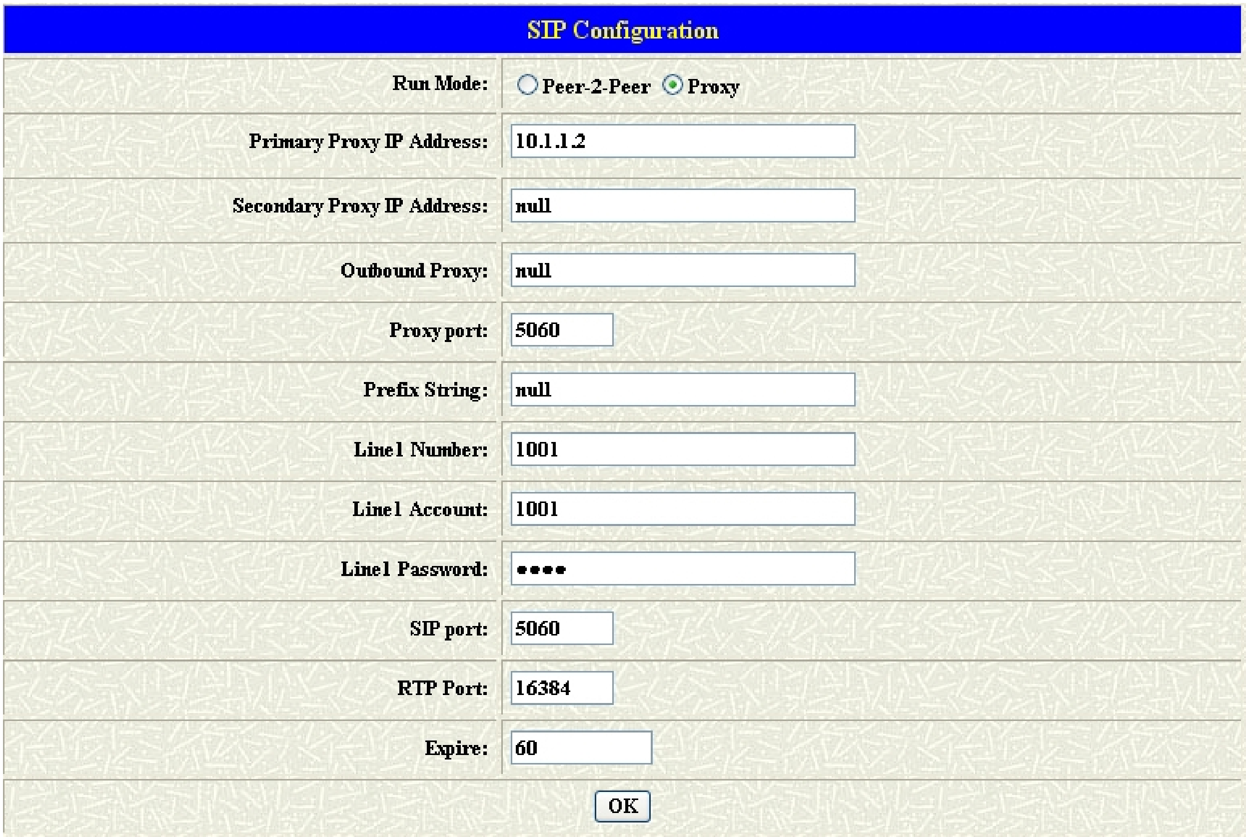
WAN IP Address When your gateway has static IP address or set behind the NAT router, configure the **WAN IP Address**, **Subnet Mask** and **Default routing gateway** together

Subnet Mask A subnet is an identifiably separate part of an organization's network. Typically, a subnet may represent all the machines at one geographic location, in one building, or on the same local area

	network (LAN). The appropriate subnet mask carried along with the packet would be: **255.255.255.0**
Default routing gateway	The default gateway IP is assigned by the ISP or your NAT router's LAN IP.
DHCP	Enable the DHCP client function if you have the Cable Modem connection to access the Internet
NAT	Network Address Translation. NAT located where the LAN meets the Internet makes all necessary IP address translations. Enable this function if your gateway is sharing the Internet connection with PC, notebook or some other network devices.
SNTP	Simple Network Time Protocol. It's a simplified version of the NTP protocol, it is an Internet protocol used to synchronize the clocks of gateway to some time reference
SNTP Server Address	Enter the preferred Time server address here You can find the public SNTP server list on Microsoft web site http://support.microsoft.com/default.aspx?scid=kb;EN-US;q262680
GMT	SNTP uses UTC(Universal Time Coordinated) as reference time, formerly and still widely called Greenwich Mean Time (GMT). Set the correct GMT for your location to get time display correctly.
IP Sharing	Enable this function when you place the gateway behind the NAT router device.
UPnP	Universal Plug and Play (UPnP) is a standard that uses Internet and Web protocols to enable devices such as PCs, peripherals, intelligent appliances, and wireless devices to be plugged into a network and automatically know about each other. With UPnP, when a user plugs a device into the network, the device will configure itself, acquire a TCP/IP address, and use a discovery protocol based on the Internet's Hypertext Transfer Protocol (HTTP) to announce its presence on the network to other devices. If the gateway behinds the NAT router with UPnP supported, you

	can enable the UPnP function.
IP Sharing Server Address	Enter the public Internet IP address here, if the gateway behinds the NAT router.
Primary DNS Server	Enter the primary Domain Name Server IP address here. If the gateway connects to the server by URL address, DNS must configure. DNS is the Domain Name System. DNS converts machine names to the IP addresses that all machines on the net have. It translates from name to address and from address to name. For example www.micronet.info
Secondary DNS Server	Enter the secondary Domain Name Server IP address here.

5.2. SIP Information



Run Mode Select Proxy mode or Peer-to-Peer mode.

Primary Proxy IP Address Set Proxy IP Address or URL e.g. **220.130.173.70** or **sip.micronet.info**

Secondary Proxy IP Address Set secondary proxy address here if available

Outbound Proxy The outbound proxy is a normal SIP proxy. You configure your client, the gateway or phone, to use the proxy for all SIP sessions, just like when you configure your Web browser to use a Web proxy for all Web transactions. In some cases, the outbound proxy is placed alongside the firewall and is the only way to let SIP traffic pass from the internal network to the Internet.
Enter the Outbound Proxy address here if your SIP service provider supported.

Proxy Port SIP local UDP port number (5060~5070), default: 5060. Change the Proxy port only when your service provider has different application.

Prefix String

Line1 Number The Line Number is same as the telephone number, people locate you by this number.

Line1 Account Account is requires by the SIP server for register, it can be the Line number, user name or e-mail account.

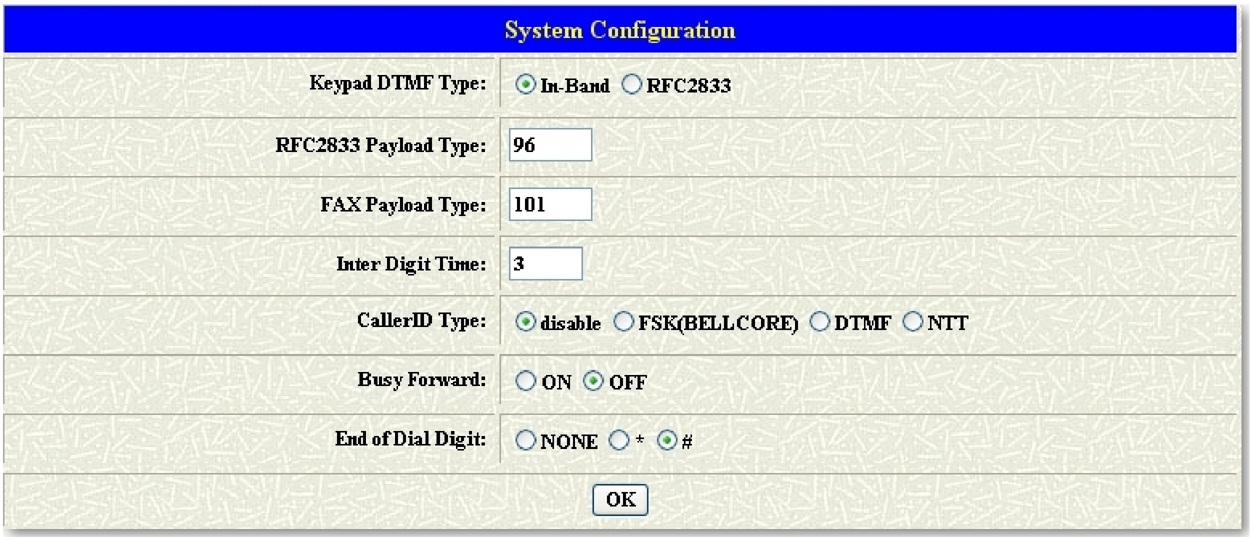
Line1 Password Enter the Account password here.

SIP Port SIP Signaling port

RTP Port Real-time Transport Protocol. The Internet protocol for transmitting real-time data such as audio and video. RTP itself does not guarantee real-time delivery of data, but it does provide mechanisms for the sending and receiving applications to support streaming data.

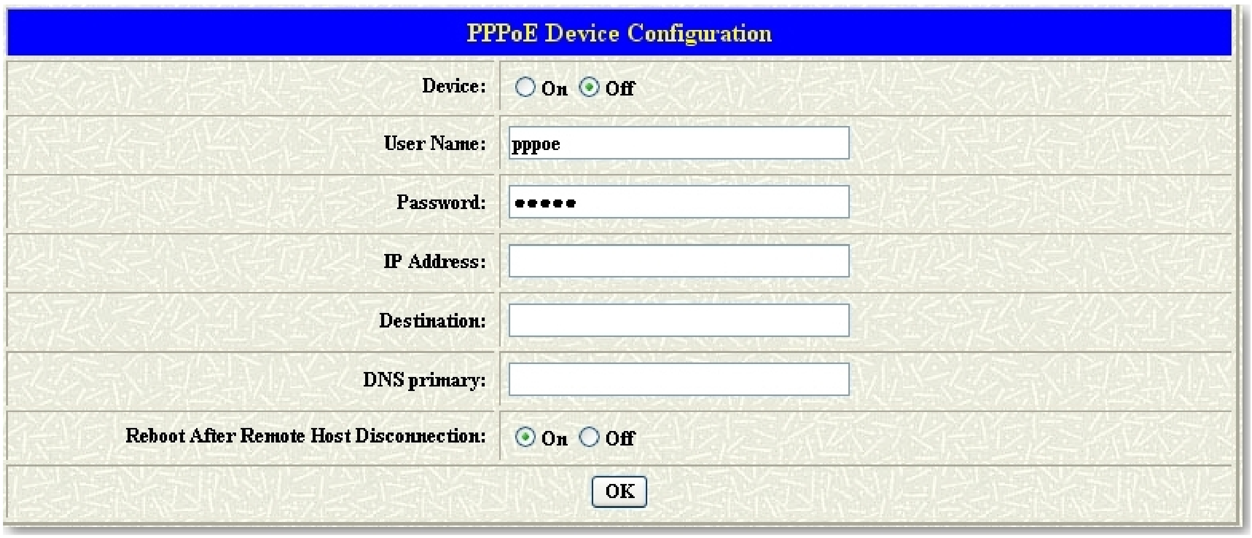
Expire Set expire time to match the SIP server registration time required. It means, if you set 60, the gateway sends the register request information to the SIP server every 60 seconds.

5.3. System Configuration



Keypad Type	Select In-Band, RFC2833 on DTMF replay type
RFC2833 Payload Type	RFC2833 Payload Type (range: 96~128 inter-used: 100, 102~105)
FAX Payload Type	Set Fax Payload Type (range: 96 or 101, default: 101)
Inter Digit Time	Set the DTMF inter digit time (second)
CallerID Type	Set CallerID type. If your telephone set has CallerID function, after the first ring at destination site, device will send line number as Caller ID to called site. FSK (Frequency-shift keying) is a method of transmitting digital signals. DTMF (dual tone multi frequency) is the signal to the phone company that you generate when you press an ordinary telephone's touch keys.
Busy Forward	Set enable or disable to route the call to preset number when the line has no answer or currently online.
End of Dial Digit	Set end of dial key as *, # , or None

5.4. PPPoE Configuration



Device Enable or Disable the PPPoE connection

User Name Enter your PPPoE account

Password Enter your PPPoE account password

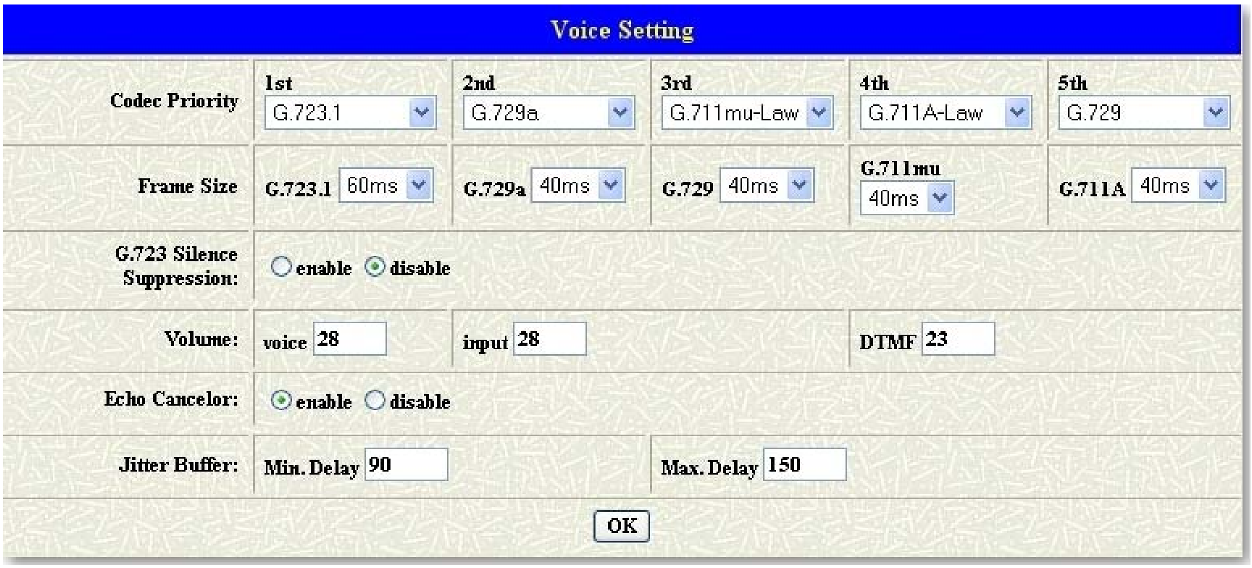
IP Address It shows the Internet connection IP address if the gateway PPPoE connection established.

Destination It shows the Internet connection gateway address if the gateway PPPoE connection established.

DNS primary It shows the Internet connection Domain Name Server IP address if the gateway PPPoE connection established.

Reboot After Remote Host Disconnection The gateway will reboot by self when lost the Internet connection and regain the connection

5.5. Voice Configuration



Codec Priority Set the Codecs priority here. If you set the g723 at first priority, g729a at second priority, then the gateway will use g723 to negotiate the connection first, then shift to second codec if the first didn't match

Frame Size Set Specify sending packet size, G.723: 30/60/90, G.711A, G.711U, G.729: 20/40/60/80ms, G.729A: 20/40/60/80ms. The smaller the packet size, the shorter the delay time. If network is in good condition, smaller sending packet size is recommended

G.723 Silence Suppression Silence Suppression, also called "Voice activation detection" (VAD) is a software application that allows a data network carrying voice traffic over the Internet to detect the absence of audio and conserve bandwidth by preventing the transmission of "silent packets" over the network.

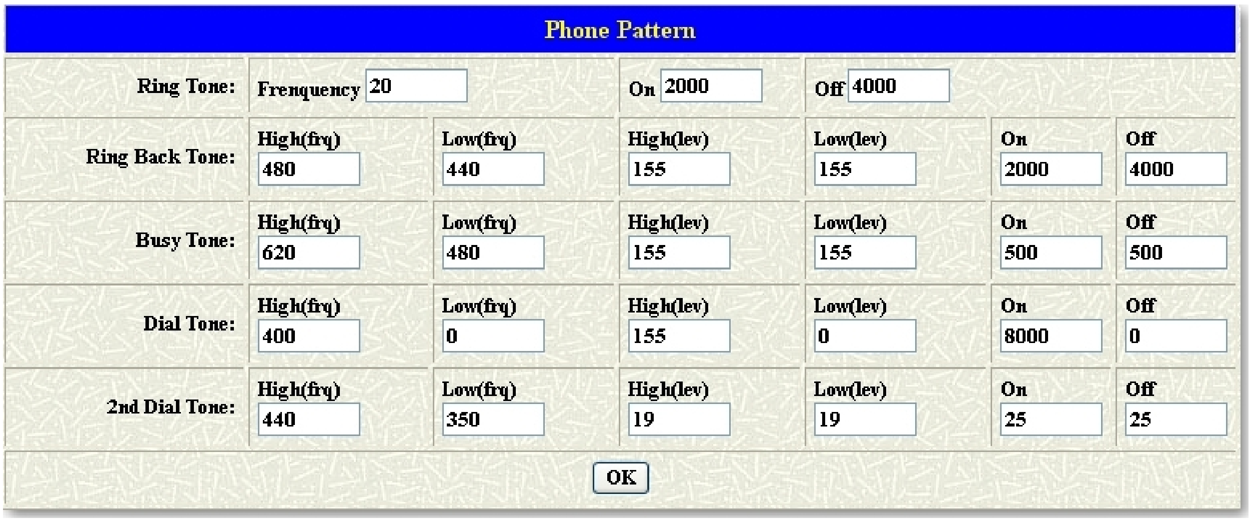
Volume Adjust the volume levels
Voice (Incoming) : 0 ~ 63
Input gain (Outgoing) : 0 ~ 38
DTMF (Keypad tone) : 0 ~ 31

Echo Cancelor Echo Canceller is designed to cancel acoustic feedback between a loudspeaker and a microphone in loud speaking audio systems.

Jitter Buffer It's a hardware device or software process that eliminates jitter caused by transmission delays in an Internet telephony (VoIP) network. As the jitter buffer receives voice packets, it adds small amounts of delay to the packets so that all of the packets appear to have been received without delays. Voice signals are sequential by nature (i.e., they must be played back in the order in which they were sent) and the jitter buffer ensures that the received packets are in the correct order. Without a jitter buffer to smooth the transmission, data can be lost, resulting in choppy audio signals.

5.6. Phone Pattern Configuration

For tone simulation, FXS Gateway adopts dual frequencies as traditional telephone does. Default tone value is set according to U.S. tone specification. Users may adjust the values to their own country's tone specification or users-defined tone specification.



Ring Tone Set Ring frequency, on time, off time. Gateway will give ring to phone set to trigger ring. If user found that phone set cannot ring when having incoming call, please try to increase ring frequency here.

- ringing frequency: 15 ~ 100 (Unit: Hz)
- ringing ring ON/OFF: 0 ~ 8000 (Unit: ms)
- ringing level: 0 ~ 94 (Unit: V)
- tone frequency: 0 ~ 65535 (Unit: Hz)
- tone freqLevel: 0 ~ 65535 (Unit: mVrms)
- tone Tone ON/OFF: 0 ~ 8000 (Unit: ms)

Ring Back Tone Set ring back tone parameters

Busy Tone Set busy tone parameters

Dial Tone Set Dial tone parameters

2nd Dial Tone To configure the value of the local 2nd dial tone

Audible tones are used in the telephone system to indicate the progress or disposition of a call. Precise dial tone consists of Current day "precise" tones consist of a summation of two low distortion sine waves.

The Dial Tone signal is used in Public Switched Telephone Networks to indicate that the telephone network switching equipment has recognized that a telephone has gone off-hook, and the switching equipment is prepared to receive the dialed digits or DTMF codes.

The Ring-back signal is used in Public Switched Telephone Networks to indicate to the caller that the called number is not busy, and that the line is being "rung" or signaled that an incoming call is present. In most cases, the ring-back signal has the same cadence as the ring generators used in that country, but the ring-back and ring generators are usually not synchronized with one another.

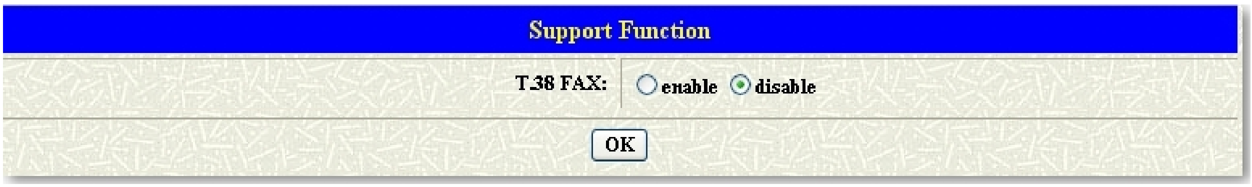
The Busy signal is used in Public Switched Telephone Networks to indicate that the called party is already taking another call. On most switching systems, the busy signal will be emitted until the caller goes on-hook.

Note:

If disconnect tone is single-frequency, user has to configure the same frequency value of “Low frequency” and “High frequency”; the same level of “Low frequency” and “High frequency”

For On/Off cadence, user must set “1023” instead of “0”, if there is only one set of cycle, please as in second set columns

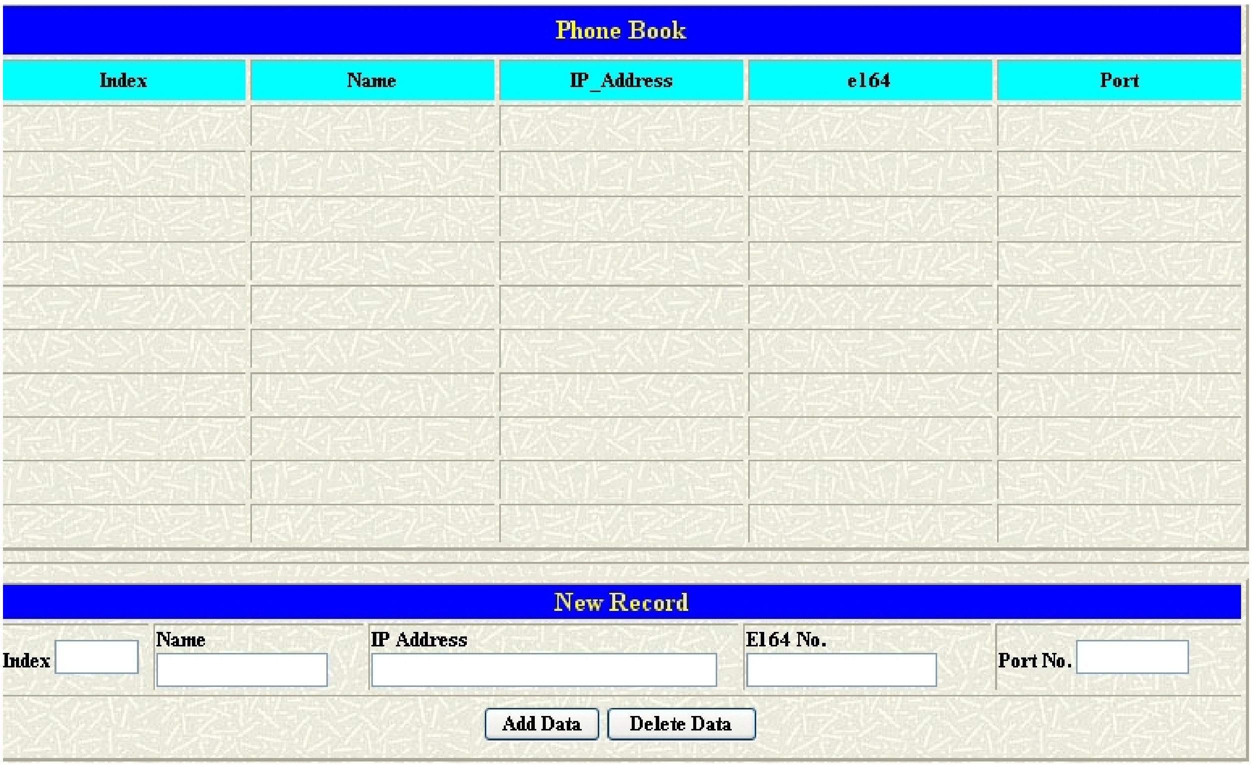
5.7. Support Configuration



T.38 FAX	T.38 is an ITU standard for sending FAX across IP networks in a real-time mode. FAX messages are sent as UDP or TCP/IP packets. Enable the T.38 FAX function, the gateway can send or receive the facsimiles. It must enable on both sides, the caller and called party.

5.8. Phone Book Configuration

Phone Book function allows users to define their own numbers, which mapping to real IP address. It is effective only in peer-to-peer mode.



Add Data You can record 20 sets of phone book. Enter the **Index**, **Name**, **IP Address** and **E.164 No** then click the Add Data to create the new phone book record.

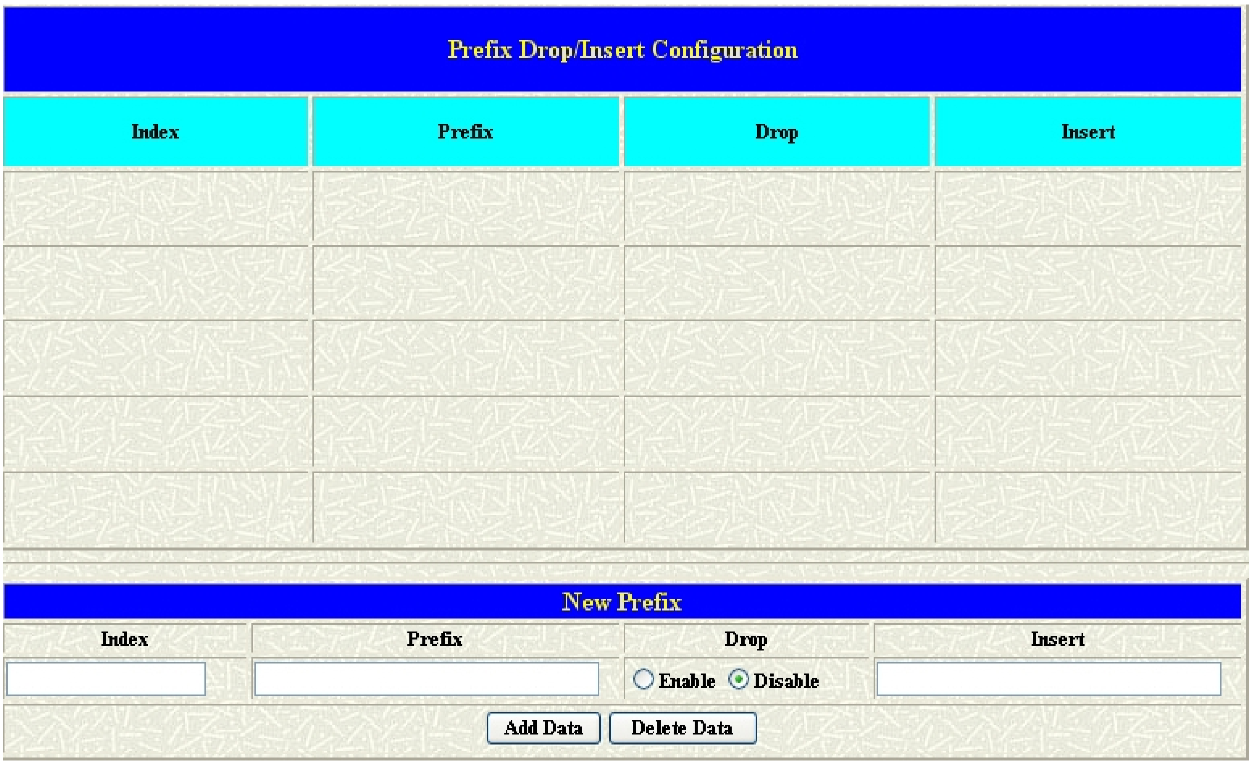
Delete Date Enter the Index and click Delete Data button to erase the phone book record.

Note:

The e164 number defined in phone book will fully carry to destination. It is not just a representative number for destination's IP Address. In other words, user dial this e164 number to reach destination, destination will receive the number and find out if it is matched to its e164, including Line number in some particular device.

5.9. Prefix Configuration

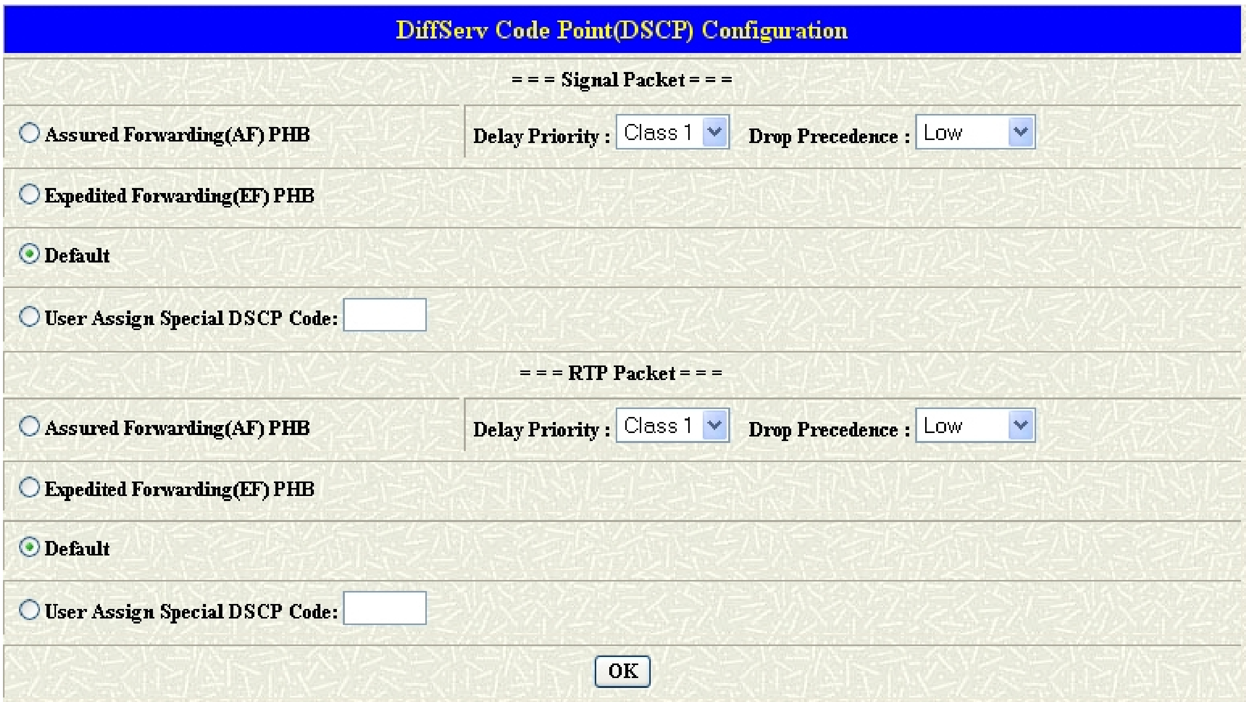
The Prefix function is using the drop and inserts digits



Add Data Enter the **Index**, **Prefix**, **Drop** enable/disable and **Insert** then click the Add Data to create the new Prefix record.

Delete Date Enter the Index and click Delete Data button to erase the Prefix record.

5.10. DSCP Configuration



Assured Forwarding (AF) PHB

Assured Forwarding (AF): Has four classes and three drop-precedence within each class (so a total of twelve codepoints). Excess AF traffic is not delivered with as high probability as the traffic "within profile," which means it may be demoted but not necessarily dropped **DiffServ AF**

Expedited Forwarding (EF) PHB

Expedited Forwarding (EF): Has a single codepoint (DiffServ value). EF minimizes delay and jitter and provides the highest level of aggregate quality of service. Any traffic that exceeds the traffic profile (which is defined by local policy) is discarded **DiffServ EF**

Default

Select TOS value as 0.

User Assign Special DSCP Code

User can set other unspecified value here

Differentiated Services (DiffServ, or DS) is a protocol for specifying and controlling network traffic by class so that certain types of traffic get precedence - for example, voice traffic, which requires a relatively uninterrupted flow of data, might get precedence over other kinds of traffic.

By using DiffServ, traffic is classified based on priority. Then the traffic is forwarded using one of three IETF-defined per-hop behavior (PHB) mechanisms. This approach allows traffic with similar service characteristics to be passed with similar traffic guarantees across multiple networks, even if the multiple networks don't provide the same service the same way. This is an important feature because the Internet is really a network of multiple service provider networks.

DiffServ replaces the first bits in the ToS byte with a differentiated services code point (DSCP). The DSCP is then mapped to the PHB. This technique allows service providers to control how the DSCP codepoints are mapped to PHBs, and each time a packet enters a network domain it may be re-marked.

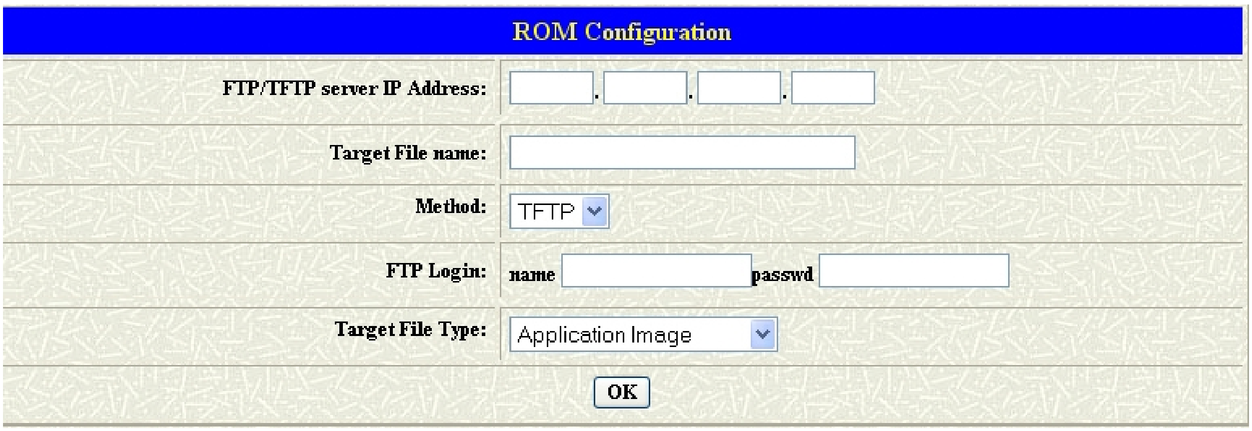
5.11. Password Configuration

Login User Name	Select root or administrator
Current Password	Enter the existing password here
New Password	Enter the new password
Confirm New Password	Enter the new password again

There is no password as default setting, it is strongly recommended that you change the factory default password of the gateway. All users who try to access the Gateway's Web-based setup menu will be prompted for the Gateway's Password. The new Password must not exceed 12 characters in length and must not include any spaces. Enter the new Password a second time to confirm it.

5.12. ROM Upgrade

The web configuration provides Update FXS Gateway ROM Version.



FTP/TFTP server IP address Enter the FTP or TFTP Server IP Address

Target File Name Enter the new firmware's file name here

Method Select download method as FTP or TFTP

FTP Login Name Enter the FTP Login name (max 14 byte)

FTP Login Password Enter the FTP Login password (max 14 byte)

Target File Type Select download Target File Type on 2M Boot Image, DSP Application Image, DSP Core Image, DSP Test Image different options from the drop-down list box

Note:

To upgrade the firmware version, use the Application ROM only in most cases. 2M ROM includes BOOT and APP images.

5.13. Flash Clean

To reset the gateway settings back to factory default

Note:

User whose login name is **root** only executes it. All configurations in **[Network Interface]** will be kept.

5.14. Commit Data

To save change after configuring FXS Gateway.

5.15. Reboot System

Reboot the FXS Gateway

6. Command List

This section introduces the command line interface and lists all of the commands. You can use the commands to configure the gateway by telnet.

Command	**Description**
help	help/man/? [command]
quit	quit/exit/close the telnet connection
debug	Show debug message
reboot	Re-start the gateway
commit	Save the change
ifaddr	Network address manipulation
time	Show current time
ping	Connection test command
pbook	Phonebook information and configuration
pppoe	PPPoE stack manipulation
flash	Clean configuration from flash rom
sysconf	System information manipulation
sip	SIP information manipulation
security	Security information manipulation
voice	Voice information manipulation
support	Special Voice function support manipulation
tos	IP Packet ToS/DSCP values
phone	Setup of call progress tones and ringing (SLIC control)
bureau	Configure the Hotline mode destination
rom	Firmware information and update
passwd	Password setting information and configuration
prefix	Prefix drop/insert information manipulation

6.1. [help]

Type [help], [man] or [?] to show the command list as the table below.

```
usr/config$ ?

help          help/man/? [command]
quit          quit/exit/close
debug         show debug message
reboot        reboot local machine
commit        commit flash rom data
ifaddr        internet address manipulation
time          show current time
ping          test that a remote host is reachable
pbook         Phonebook information and configuration
pppoe         PPPoE stack manipulation
flash         clean configuration from flash rom
sysconf       System information manipulation
sip           SIP information manipulation
security      Security information manipulation
voice         Voice information manipulation
support       Special Voice function support manipulation
tos           IP Packet ToS/DSCP values
phone         Setup of call progress tones and ringing (SLIC control)
bureau        Bureau line information manipulation
rom           ROM file update
passwd        Password setting information and configuration
prefix        Prefix drop/insert information manipulation
```

6.2. [quit]

Type [quit] will quit and disconnect the Gateway configuration mode.

6.3. [debug]

Open debug message will show up specific information while Gateway is in operation. After executing the debug command, it should execute command [debug -open] as well.

```
usr/config$ debug

Debug message information and configuration
Usage:
debug [-add type1 [[type2]...]] | -open | -close | -status

    -status    Display the enabled debug flags.
    -add       Add debug flag.
    -delete    Remove specified debug flag.
    -open      Start to show debug messages.
    -close     Stop showing debug messages.
Example:
    debug -add sip msg
    debug –open
```

Parameter Usage:

-status	Display the enabled debug flags.
-add	Add debug flag
-sip	SIP related information
-msg	voice related information
-delete	Remove specified debug flag
-open	Start to show debug messages
-close	Stop showing debug messages

For example, user open debug flags including sip, vp, msg.

```
usr/config$ debug -add sip msg
usr/config$ debug -open
```

```
usr/config$ debug -status

Current debug type enabled :
Debug Mode is open
DEBUG-> SIP    MSG
```

6.4. [reboot]

```
usr/config$ reboot

♦  Rebooting...It will take 40 seconds....
```

After [commit] command, type [reboot] to re-start the gateway to take new configurations effective

6.5. [commit]

```
usr/config$ commit

This may take a few seconds, please wait....

Commit to flash memory ok!
```

Save changes after configuring Gateway.

6.6. [ifaddr]

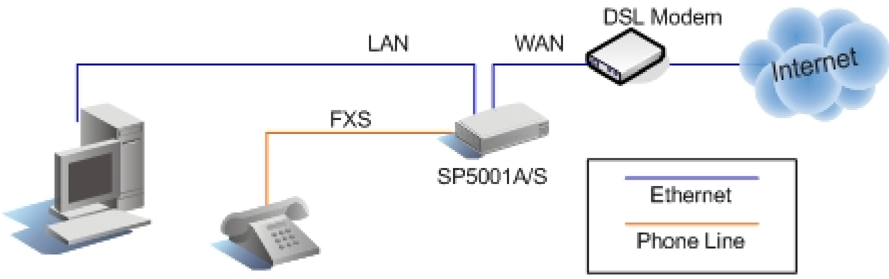
Configure and display Gateway network information.

```
usr/config$ ifaddr

LAN information and configuration
Usage:
ifaddr [-print]|[-dhcp used]|[-sntp mode [server]]
ifaddr [-ip ipaddress] [-mask subnetmask] [-gate defaultgateway]
ifaddr [-dns index [dns server address]] [-ipsharing used[ip address]]

   -print    Display LAN information and configuration.
   -ip       Specify WAN ip address.
   -lanip    Specify LAN ip address.
   -mask     Set Internet subnet mask.
   -gate     Specify default gateway ip address
   -nat      Set NAT service flag (On/Off).
   -dhcp     Set DHCP client service flag (On/Off).
   -sntp     Set SNTP server mode and specify IP address.
   -dns      specify IP address of DNS Server.
   -timezone  Set local timezone.
   -ipsharing Specify usage of an IP sharing device and specify IP address.
   -server   specify EMS Server IP address
   -id       specify EMS Server ID
   -pwd      specify EMS Server password
   -emstime  specify EMS cycle time
Note:
   Range of ip address setting (0.0.0.0 ~ 255.255.255.255).
   DHCP client setting value (On=1, Off=0). If DHCP set to 'On',
   Obtain a set of Internet configuration from DHCP server assigned.
   SNTP mode (0=no update, 1=specify server IP, 2=broadcast mode).
Example:
   ifaddr -ip 210.59.163.202 -mask 255.255.255.0 -gate 210.59.163.254
   ifaddr -nat 1
   ifaddr -dhcp 1
   ifaddr -sntp 1 210.59.163.254
   ifaddr -ipsharing 1 210.59.163.254
   ifaddr -dns 1 168.95.1.1
```

Parameter Usage:

-print	Print current IP setting and status
-ip	Assign the VoIP gateway's IP address
-lanip	Specify LAN port IP address (For NAT function), use this command setup lanip address assigned to PC or other machine.

	Setting IP address provide PC setup Default Gateway Address
-mask	Assign the VoIP gateway's Subnet Mask
-gate	Assign the VoIP gateway's default gateway
-nat	Provides Network Address Translation function



Enable the NAT function when share the connection to your PCs.

-dhcp	Dynamic host configuration (0=Off, 1=On)
-sntp	Simple Network Time Protocol (0=No update, 1=Specify server IP). When SNTP function is activated, users have to specify a SNTP server as network time source Example : **ifaddr -sntp 1 168.95.192.12**
-dns	Specify the DNS server's IP address
-timezone	Set local time zone according to GMT
-ipsharing	Enable this function when the VoIP gateway behind the NAT router or IP Sharing devices. Example : **ifaddr -ipsharing 1 61.219.198.204**

Note : If you don't have static public IP address, then the dedicated IP address is not necessary in the command, for example : **ifaddr -ipsharing 1** However, dynamic IP Address is not working in Peer-to-Peer mode.

-server	specify EMS Server IP address
-id	specify EMS Server ID
-pwd	specify EMS Server password
-emstime	specify EMS cycle time

The EMS (Element Management System) is expressly built to simplify deployment, configuration and management of network equipment and to help you streamline delivery of the high-demand services and capabilities enabled.

Note:

One Group only use only LAN IP address, if have two gateway on this group, you must change second gateway LAN IP Address different first gateway.

```
Gateway First:
usr/config$ ifaddr -lanip 192.168.124.124

Gateway Second:
usr/config$ ifaddr -lanip 192.168.124.125
```

Information Example:

```
usr/config$ ifaddr -print

Internet address information
   WAN IP address      : 192.168.0.243
   Subnet mask         : 255.255.255.0
   Default gateway     : 192.168.0.1
   NAT enabled         : OFF
   DHCP startup        : OFF
   SNTP                : mode=1
                         server 168.95.195.12
                         time zone : GMT+8
                         cycle=1024 mins

   IPSharing           : no IPSharing device.

   Primary DNS Server   : 168.95.1.1
   Secondary DNS Server : 168.95.1.1

   EMS IP Address      : 192.168.1.1
   EMS User ID         : vwusr
   EMS Password        : vwusr
   EMS cycle time      : 0
```

6.7. [time]

When SNTP function of Gateway is enabled and SNTP server can be found as well, type [time] command to show current network time.

```
usr/config$ time
Current time is WED DEC 01 12:38:38 2004
```

6.8. [ping]

ping is the name of a computer network tool used on TCP/IP networks (such as the Internet). It provides a basic test of whether a particular host is operating properly and is reachable on the network from the testing host. It works by sending ICMP packets to the target host and listening for replies

For example: if 192.168.1.2 is not existing while 192.168.123.100 exists. Users will have the following results:

```
usr/config$ ping 192.168.1.2

PING 192.168.1.2: 56 data bytes
no answer from 192.168.1.2
```

```
usr/config$ ping 192.168.123.100

PING 192.168.123.100: 56 data bytes
64 bytes from 192.168.123.100: icmp_seq=0. time=5. ms
64 bytes from 192.168.123.100: icmp_seq=1. time=0. ms
64 bytes from 192.168.123.100: icmp_seq=2. time=0. ms
64 bytes from 192.168.123.100: icmp_seq=3. time=0. ms
----192.168.123.100 PING Statistics----
4 packets transmitted, 4 packets received, 0% packet loss
round-trip (ms)  min/avg/max = 0/1/5
```

6.9. [pbook]

Phone Book function allows users to define their own numbers, which mapping to real IP address. It is effective only in **peer-to-peer mode**. When adding a record to Phone Book, users do not have to reboot the machine, and the record will be effective immediately.

```
usr/config$ pbook

Phonebook information and configuration
Usage:
pbook [-print [start_record] [end_record]]
pbook [-add [ip ipaddress] [name Alias] [e164 phonenumber]]
pbook [-search [ip ipaddress] [name Alias] [e164 phonenumber]]
pbook [-insert [index] [ip ipaddress] [name Alias] [e164 phonenumber] [port numb
er]]
pbook [-delete index]
pbook [-modify [index] [ip ipaddress] [name Alias] [e164 phonenumber] [port numb
er]]

    -print    Display phonebook data.
    -add      Add an record to phonebook.
    -search   Search an record in phonebook.
    -delete   Delete an record from phonebook.
    -insert   Insert an record to phonebook in specified position.
    -modify   Modify an exist record.
Note:
    If parameter 'end_record' is omited, only record 'start_record' will be disp
lay.
    If  both parameters 'end_record' and 'start_record' are omited, all records
will be display.
    Range of ip address setting (0.0.0.0 ~ 255.255.255.255).
    Range of index setting value (1~100),
Example:
    pbook -print 1 10
    pbook -print 1
    pbook -print
    pbook -add name Test ip 210.59.163.202 e164 1001
    pbook -insert 3 name Test ip 210.59.163.202 e164 1001
    pbook -delete 3
    pbook -search ip 192.168.4.99
    pbook -modify 3 name Test ip 210.59.163.202 e164 1001
```

Parameter Usage:

-print Print out current contents of Phone Book. Users can also add index number, from 1 to 50, to the parameter to show specific phone number.

-add	add a new record to phone book. When adding a record, users have to specify name, IP, and e164 number to complete the command.
name	Name to represent caller.
e164	E.164 number for mapping with IP address of caller
ip	IP address of caller
port	Call signal port number of caller
drop	Drop e.164 number when dial out. 0 means to keep e.164 number, 1 means to drop e.164 number when dialing out.
insert	Insert digits.(1~10 digits)
-modify	modify an existing record. When using this command, users have to specify the record's index number, and then make the change.
-delete	delete a specific record. For example : pbook -delete 3

Note:

Index number: means the sequence number in phone book. If users do request a specific index number in phone book, Gateway will give each record a automatic sequence number as index.

PhoneBook Rules:

The e164 number defined in phone book will fully carry to destination. It is not just a representative number for destination's IP Address. In other words, user dial this e164 number to reach destination, destination will receive the number and find out if it is matched to its e164, including Line number in some particular device.

For example:

```
usr/config$ pbook -print

index   Name          IP                  E164           Port
======================================================================
1      SP5100        192.168.0.242       5100
----------------------------------------------------------------------
```

6.10. [pppoe]

Display PPPoE related information.

```
usr/config$ pppoe

PPPoE device information and configuration
Usage:
pppoe [-print]|[-open]|[-close]
pppoe [-dev on/off][-id username][-pwd password][-reboot on/off]

    -print     Display PPPoE device information.
    -dev       Enable(=1) or Disable(=0) device.
    -open      Open PPPoE connection.
    -close     Disconnect PPPoE connection.
    -id        Connection user name.
    -pwd       Connection password.
    -reboot    Reboot after remote host disconnection.
```

Parameter Usage:

-print	print PPPoE status.
-dev	Enable or Disable PPPoE Dial-up function
-open	Open the connection
-close	Disconnect the connection
-id	The User name ID provided by ISP
-pwd	The Login password provided by ISP
-reboot	Reboot the gateway after the PPPoE connection disconnected

6.11. [flash]

Restore the gateway's configurations back to default.

```
usr/config$ flash

Flash memory information and configuration
Usage:
flash -clean

Note:
    This command will clean the configuration stored in
    the flash and reboot it.
```

Parameter Usage:

-clean	clean all the user defined value, and reboot Gateway in factory default mode

Note:

It is recommended that use [flash -clean] after application firmware upgraded.

User whose login name is root only executes it. All configurations in command [ifaddr] and [pppoe] will be kept.

6.12. [sysconf]

This command displays system information and configurations.

```
usr/config$ sysconf

System information and configuration
Usage:
 sysconf [-print] [-idtime digit] [-bf digit] [-keypad dtmf]
        [-faxtype type][-2833type type][-lcdrop ON/OFF]
        [-droptime digit][-eod digit] [-callerid type]
        [-service used][-dtmfstart digits] [-dtmfend digits]
 sysconf -print

 -print        Display system overall information and configuration.
 -idtime       Inter-Digits time.(1~10 sec)
 -service      Specify gateway service type. (0: Dial in service,
               1: HotLine service.)
 -keypad       Select DTMF type: 0=In-band,
                              1=RFC2833.
 -faxtype      FAX Payload Type    (range:96~128 inter-used:100,102~105)
 -2833type     RFC2833 Payload Type (range:96~128 inter-used:100,102~105)
 -lcdrop       Disconnect Supervision(Loop Current Drop) (ON:1 / OFF:0)
 -droptime     Period of Loop Current Drop (ms)
 -eod          End of Dial Digit setting(0: none, 1: *, 2: #)
 -callerid     Caller ID Type setting, 0: Disable,
                                   1: FSK(BELLCORE),
                                   2: DTMF,
                                   3: NTT.
 -dtmfstart    DTMF CallerID Start Symbol.
 -dtmfend      DTMF CallerID End Symbol.
Example:
  sysconf -keypad 0 -eod 2 -callerid 1
```

Parameter Usage:

-print	Show the sysconf current status.
-idtime	Set the duration (in second) of two pressed digits in dial mode as timed out. If after the duration user hasn't pressed next number, it will dial out all number pressed (1-10 seconds).
-service	Specify gateway service type. (0: Dial in service, 1: HotLine service.)
-keypad	DTMF replay type. When value is “1”, FXS Gateway will transfer DTMF signal via RTP payload as defined in RFC2833. When the value is set to “0”, the DTMF type is set as In-band.
-faxtype	FAX Payload Type. Rrange:96~128 inter-used:100,102~105.
-2833type	RFC2833 Payload Type. Range: 96~128 inter-used: 100, 102~105.
-lcdrop	Disconnect Supervision (Loop Current Drop) (ON:1 / OFF:0).

-droptime	Period of Loop Current Drop (ms).
-eod	Select the End of Dial key, “#”, “*” or none
-callerid	Select the Caller ID type, 0 = disable, 1 = FSK(Bell core), 2 = DTMF, 3 = NTT. After the first ring at destination site, device will send line number as caller ID to called site.
-dtmfstart	DTMF Caller ID Start Symbol
-dtmfend	DTMF Caller ID End Symbol

Payload Type, the essential data that is being carried within a packet or other transmission unit. The payload does not include the "overhead" data required to get the packet to its destination. Note that what constitutes the payload may depend on the point-of-view. To a communications layer that needs some of the overhead data to do its job, the payload is sometimes considered to include the part of the overhead data that this layer handles. However, in more general usage, the payload is the bits that get delivered to the end user at the destination.

6.13. [sip]

This command is to configure SIP related parameters.

```
usr/config$ sip

SIP stack information and configuration
Usage:
sip [-print] [-mode  pxmode] [-outpx IPaddmress][-transport type]
sip [-px address] [-px2 address] [-pxport number] [-prefix prefixstring]
   [-line1 number]
   [-expire t1] [-port udpPort] [-rtp rtpPort]
sip -print

   -print    Display SIP stack information and configuration.
   -mode     Configure as Peer-to-Peer mode:0/Proxy mode:1.
   -px       Primary Proxy server address. (IPv4 address or dns name)
   -px2      Secondary Proxy server address. (IPv4 address or dns name)
   -pxport   Proxy server port.   (the port of proxy)
   -outpx    OutBound Proxy server address. (IPv4 address or dns name)
   -prefix   Specify prefix string, use it when UserID contains alphabets
            (if UserID uses numerals, specify as null)
   -line1    TEL1 Phone number.
   -pbsearch  Search phone book   0:off/1:on.
   -expire   The relative time after which the message expires(0 ~ (2^31-1))
   -port     SIP local UDP port number (5060~5070), Default: 5060
   -rtp      RTP port number (2326~65534), Default: 16384
Example:
    sip -mode 1
    sip -px 210.59.163.171 -line1 70
```

Parameter Usage:

-print	Show the SIP current settings
-mode	Select the P2P mode or Proxy mode, 0 = P2P, 1 = Proxy
-px	To specify Proxy address when FXS Gateway is in proxy mode. Proxy address can be IPv4 address or DNS name.
-px2	To specify Secondary Proxy server address.
-pxport	To configure proxy server signaling port, default value is 5060, if there is no special request of Proxy server, please don’t change this value.
-outpx	Set IP Address or URL address (Domain Name Server must be configured. Please refer to Network Configure) of outbound Proxy server.
-prefix	when your username contains alphabets, for example sip1123, then specify the prefix string as “sip”.
-line1	Assign gateway’s line number
-pbsearch	enable/disable phone book search function under Proxy Mode. If

	user enabled this function, the gateway will search dialed number in phone book to see if there is any matched table before send to Proxy server, and if there is a matched data in phone book, the gateway will make call to related IP address.
-expire	This parameter set duration time for sending registration information.
-port	SIP port which used to listen incoming SIP messages
-rtp	Specify the RTP received port number

6.14. [security]

This command is used to configure the account information included username and password obtained from the proxy service provider

```
usr/config$ security

Security information and configuration
Usage:
 security [-line number][-name username] [-pwd password]
 security [-print]

 -print       Display system account information and configuration.
 -line        Specify which line number you want to set the account.
 -name        Specify user name.
 -pwd         Specify password.
Example:
  security -line 1 -name 1001 -pwd 1001
```

Parameter Usage:

-print	Shows the current settings
-line	Specify the line for the account configuration, here has only one line for this gateway model.
-name	Specify the username of your account information.
-pwd	Specify the password of your account information.

6.15. [voice]

The voice command is associated with the audio setting information.

```
usr/config$ voice

Voice codec setting information and configuration
Usage:
voice [-send [G723 ms] [G711U ms] [G711A ms] [G729 ms] ]
     [-volume [voice level] [input level] [dtmf  level]]
     [-nscng [G711U used1] [G711A used2] [G723 used3]]
     [-echo used] [-mindelay t1] [-maxdelay t2]
voice -print
voice -priority [G723] [G711U] [G711A] [G729]

   -print    Display voice codec information and configuration.
   -send     Specify sending packet size.
            G.723  (30/60 ms)
            G.711U (20/40/60 ms)
            G.711A (20/40/60 ms)
            G.729  (20/40/60/80 ms)
   -priority  Priority preference of installed codecs.
            G.723
            G.711U
            G.711A
            G.729
   -volume   Specify the following levels:
            voice volume (0~63, default: 25),
            input gain (0~38, default: 25),
            dtmf  volume (0~31, default: 23),
   -nscng    No sound compression and CNG. (G.723.1 only, On=1, Off=0).
   -echo     Setting of echo canceller. (On=1, Off=0, per port basis).
   -mindelay  Setting of jitter buffer min delay. (0~150, default: 90).
   -maxdelay  Setting of jitter buffer max delay. (0~150, default: 150).
Example:
   voice -send g723 60 g711u 60 g711a 60 g729 60
   voice -volume voice 20 input 32 dtmf 27
   voice -echo 1
```

Parameter Usage:

-print Shows the current settings

-send To define packet size for each codec. 20/40/60/80 ms means to send a voice packet per 20/40/60/80 milliseconds. The smaller the packet size, the shorter the delay time. If network is in good condition, smaller sending packet size is recommended. In this parameter, 20/40/60ms is applicable to G.711u/a law, 20/40/60ms is applicable to G.729 codec, while 30/60ms is applicable to G.723.1 codec.

-priority	Codec priority while negotiating with other SIP device. The codec listed in left side has the highest priority when both parties determining final codec. For example : usr/config$ voice -priority g729 g723 g711u g711a (Selected four Codecs, G.729 is the first choice)
-volume	To adjust the voice, input and dtmf levels
voice	which can be heard from Gateway side(range 0~63, default: 25).
input	which the opposite party hears (range 0~38, default: 25).
dtmf	which sends to its own Line (range 0~31, default: 23).
-nscng	Silence suppression and comfort noise generation setting (1 = ON; 0 = OFF). It is applicable to G.723 codec only.
-echo	Enable or Disable the echo cancellation
-mindelay	The minimum jitter buffer size (Default value= 90 ms).
-maxdelay	The minimum jitter buffer size (Default value= 150 ms).

Note:

Be sure to know well the application before you change voice parameters because this might cause incompatibility.

6.16. [support]

This command provides some extra functions that might be needed by users.

```
usr/config$ support

Special Voice function support manipulation
Usage:
support [-t38 enable]
       [-busy number] [-noanswer number] [-uncon number]
support -print
   -t38      T.38(FAX) enabled/disabled.
   -busy     Busy Forward number.(if empty, please fill "null")
   -noanswer No Answer Forward number.(if empty, please fill "null")
   -uncon    Unconditional Forward number.(if empty, please fill "null")
Example:
   support  -t38 1
   support  -busy 1001
   support  -uncon null
```

Parameter Usage:

-print	Shows the current settings
-t38	Enable or disable FAX ability. The function is will automatically defer codec (G.723 or G.729a) to T.38 when FAX signal is detected.
-busy	Provide setting busy forward to other number, when your gateway is setting this function, it will forward to setting phone number if the channel is busy,
-noanswer	Provide setting noanswer forward to other number, when you set this function, it will forward to setting phone number if no one answer the call.
-uncon	Provide setting Unconditional forward to other number, when you set this function, all the calls to your number will forward to setting phone number.

6.17. [tos]

IP Packet ToS (Type of Service)/ Differentiated Service configuration.

```
usr/config$ tos

IP Packet ToS(type of Service)/Differentiated Service configuration
Usage:
tos [-rtptype dscp]
tos [-sigtype dscp]
tos -print
    [-rtpreliab mode]
tos -print

Example:
   tos -rtptype 7 -sigtype 0
```

-rtptype the packages of voice (0~63)

-sigtype the package of call signal (0~63)

IPv4 Head Format

+	0 - 3	4 - 7	8 - 15	16 - 18	19 - 31
0	Version	Header length	Type of Service (now DiffServ and ECN)	Total Length	
32	Identification			Flags	Fragment Offset
64	Time to Live		Protocol	Header Checksum	
96	Source Address				
128	Destination Address				
160	Options				
192	Data				

In RFC 791, the following 8 bits were allocated to a Type of Service (ToS) field - now DiffServ and ECN. For instance, one host could set its IPv4 datagrams' ToS field value to prefer low delay, while another might prefer high reliability. In practice, the ToS field has not been widely implemented. However, a great deal of experimental, research and deployment work has focused on how to make use of these eight bits. These bits have been redefined and most recently through DiffServ working group in the IETF and the Explicit Congestion Notification codepoints

Note:

The value of rtptype and sigtype is from 0 to 63. ToS only works if it has related network devices supported.

6.18. [phone]

Gateway progress tone is configurable. Default tone value is set according to U.S. tone specification. Users may adjust the values according to their own country's tone specification or users-defined tone specification.

```
usr/config$ phone

Phone ringing , ringback tone , busy tone , dial tone setting and notes
Usage:

phone [-ring [freq   ] [ringON  ] [ringOFF ] [ringLevel]]
     [-rbt  [freqHi ] [freqLo  ] [freqHiLev] [freqLoLev]
           [Tone1ON] [Tone1OFF] [Tone2ON  ] [Tone2OFF ]]
     [-bt   [freqHi ] [freqLo  ] [freqHiLev] [freqLoLev]
           [Tone1ON] [Tone1OFF] [Tone2ON  ] [Tone2OFF ]]
     [-dt   [freqHi ] [freqLo  ] [freqHiLev] [freqLoLev]
           [Tone1ON] [Tone1OFF] [Tone2ON  ] [Tone2OFF ]]
     [-flash [freqLo ] [freqHi ]]
     [-level [loopCurrentLevel] [onhookLineVoltageLevel ]]
phone [-print [ring]|[rbt]|[bt]|[dt]|[flash]]

     -print  Display phone ringing/tone configuration.
            ring :   ringing
            rbt  :   ringback tone
            bt   :   busy tone
            dt   :   dial tone
            flash:   flash tone
     -ring   ringing configuration set .
     -rbt    ringback tone configuration set .
     -bt     busy tone configuration set .
     -dt     dial tone configuration set .
     -flash  flash configuration set .
     -level  Loop Current and On-Hook Line Voltage level set .
Note:
     ringing frequency   : 15 ~ 100   (Unit : Hz)
     ringing ring ON/OFF : 0  ~ 8000  (Unit : ms)
     ringing level      : 0  ~ 94    (Unit : V)
     tone    frequency   : 0  ~ 65535 (Unit : Hz)
     tone    freqLevel   : 0  ~ 65535 (Unit : mVrms)
     tone    Tone ON/OFF : 0  ~ 8000  (Unit : ms)
     level   loopCurrent : 0  ~ 7     (20mA ~  41mA, Step :  3mA)
     level   OnHookVol   : 0  ~ 63    (  0V ~ 94.5V, Step : 1.5V)
Example:
     phone -print rbt
     phone -ring 20 2000 4000 94
     phone -rbt 480 440 125 105 2000 4000 2000 4000
     phone -bt 620 480 125 105 500 500 500 500
     phone -dt 440 350 96 96 8000 0 8000 0
     phone -flash 400 800
     phone -level 1 32
```

Parameter Usage:

-print	Specify which tone settings you want to display **ring** : ring tone settings **rbt** : ring back tone settings **bt** : busy tone settings **dt** : dial tone settings **flash** : flash time settings
-ring	To set RING tone value. The played tone type, when Gateway is receiving a call.
-rbt	To set Ring Back Tone value. The played tone type, when Gateway receives a Q.931 Alerting message. In condition that Gateway is the originate side.
-bt	To set Busy Tone value. The played tone type, when destination is busy.
-dt	To set Dial Tone value. The played tone type, when hook off a phone set of workable Gateway.
-flash	Set the detective flash range in ms, for example, 400-800 ms.
-level	Loop Current and On-Hook Line Voltage level set.

Note:

For tone simulation, Gateway adopts dual frequencies as traditional telephone does. If users want to have their own call progress tone, they can change the value of tones. High and Low frequency/level/cadence can be configured respectively.

6.19. [bureau]

To set Hotline function must be under Peer-to-Peer mode and switch to hotline mode.

```
usr/config$ bureau

Bureau line setting information and configuration
Usage:
bureau  [-hotline [Port DestIP TELnum]]
bureau -print

   -print    Display Bureau line information and configuration.
   -hotline   Set Hot line information. (Port range: 1~6)
Note:
   Hotline feature should be used together with:
      $sysconf -service 1 (HotLine service)
      $sip   -mode   0 (peer-to-peer mode)
Example:
   bureau -hotline 1 192.168.4.69 628
```

Parameter Usage:

-print	Shows the current settings
-hotline	Define Line Hotline table respectively. The table is included [Line number], [destination IP Address] and [destination Port or Number].

For example

1. Destination is a FXS device, 628 is its Line1 number

usr/config$ bureau -hotline 1 200.168.4.69 628

User picks up the telephone handset connects to gateway, and then hears the ringback tone generated from destination. Of course, the destination line 628 is ringing simultaneously.

2. Destination is a FXO device, Port_1 has connected to PSTN Line.

usr/config$ bureau -hotline 1 200.168.4.69 82265699

User picks up the Line1, and then hears the ringback tone generated from destination. Simultaneously, 82265699 numbers is the destination, which is dialed from Port_1 (Above FXO example is subject to the FXO configurations, such as 2nd dial ON or OFF.)

6.20. [rom]

ROM file information and firmware upgrade function.

```
usr/config$ rom

ROM files updating commands
Usage:
rom [-print] [-app] [-boot] [-dsptest] [-dspcore] [-dspapp]
   [-ht] [-method used] [-boot2m]
   -s TFTP/FTP server ip -f filename
rom -print
   -print    show versions of rom files. (optional)
   -app      update main application code(optional)
   -boot     update main boot code(optional)
   -boot2m   update 2M code(optional)
   -ht       updata Hold Tone PCM file(optional)
   -dsptest  update DSP testing code(optional)
   -dspcore  update DSP kernel code(optional)
   -dspapp   update DSP application code(optional)
   -s        IP address of TFTP/FTP server (mandatory)
   -f        file name(mandatory)
   -method   download via TFTP/FTP (TFTP: mode=0, FTP: mode=1)
   -ftp      specify username and password for FTP
Note:
   This command can run select one option in 'app', 'boot',
   , 'dsptest', 'dspcore', and 'dspapp'.
Example:
   rom -method 1
   rom -ftp vwusr vwusr
   rom -app -s 192.168.4.101 -f app.bin
```

Parameter Usage:

-print	Shows the current settings
-app	update application program code
-boot	update boot code
-boot2m	Includes APP and Boot code
-ht	updata Hold Tone PCM file(optional)
-dsptest	update DSP testing code(optional)
-dspcore	update DSP kernel code(optional)
-dspapp	update DSP application code(optional)
-s	To specify TFTP / FTP server IP address for upgrading
-f	To specify the target file name, this will replace the old one.
-method	To decide using TFTP or FTP as file transfer server. TFTP = 0 , FTP = 1
-ftp	If users choose FTP in above item, it is necessary to specify pre-defined username and password when upgrading files.

6.21. [passwd]

For security concern, users have to input the password before entering configuration mode. [passwd] command is for password setting purpose.

```
usr/config$ passwd

Password setting information and configuration
 Usage:
   passwd -set Loginname Password
   passwd -clean
 Note:
   1. Loginname can be only 'root' or 'administrator'
   2. passwd -clean will clear all passwd stored in flash,
      please use it with care.
 Example:
   passwd -set root Your_Passwd_Setting
```

Parameter Usage:

-set	Set login name and password, input login name then input new password.
-clean	Clear all password setup, and change null.

Note:

Gateway Login name only use **root** or **administrator**. Both accounts have the same authorization, except commands that can be executed by login name **root** only [passwd -set root], [rom -boot], [room -boot2m] and [flash -clean].

6.22. [prefix]

Prefix drop/insert information manipulation

```
Prefix drop/insert information and configuration
Usage:
prefix -add [prefix number][drop number][insert digits]
prefix -delete index
prefix -modify index [prefix number][drop number][insert number]
prefix -print   Prefix drop/insert information.
      prefix   The prefix of dialed number.
      drop     Drop prefix(Enable:1/Disable:0).
      insert   Insert digits.
Example:
    prefix -add prefix 100 drop 1 insert 2000
    prefix -add prefix 100 drop 1
    prefix -add prefix 100 drop 0 insert 200
    prefix -delete 1
    prefix -modify 1 prefix 100 drop 0 insert 300
```

Parameter Usage:

-add Add a rule to drop or insert prefix digits of incoming call.

prefix : Set which prefix number to implement prefix rule.

drop : Enable or disable drop function. If this function is enabled, Gateway will drop prefix number on incoming call.

insert : Set which digit to insert on incoming call.

-modify Modify a rule to drop or insert prefix digits of incoming call.

-delete Delete a rule to drop or insert prefix digits of incoming call.